PHD

A local area network for digitized speech in advanced aircraft communication systems

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A LOCAL AREA NETWORK

FOR DIGITIZED SPEECH IN ADVANCED

AIRCRAFT COMMUNICATION SYSTEMS

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SUMMARY

The idea for a new local area network, referred to as the Russell Ring, was a result of a requirement for a new generation of digital communication control systems for aircraft.

This thesis examines the design considerations which gave rise to this new network. It investigates in particular the network distribution method and the requirement for carrying voice signals in digital multiplexed intra-aircraft communications. The system is capable of integrating voice with other digital data, as well as providing control of radios, distribution, selection, isolation, coding, summation and amplification of signals.

The ring protocol is examined and compared with those of other well known local area networks. Methods of assessing and comparing audio quality are investigated and a new Digital Speech Quality Index proposed. A practical solution is proposed which takes advantage of recent advances in technology enhancing TEMPEST and EMC capability.

The solution is shown to have improved integrity and reliability, enhanced fault tolerance and better expandability as well as meeting the requirement for simultaneous reception of multiple conversations in a high noise environment.

CONTENTS

1. INTRODUCTION

This thesis investigates and discusses particular areas of technology and its implementation in a system consisting of a range of inter-related technologies. The requirement for digital voice on a local area network in an intra-aircraft environment, is a combination not previously researched.

The basic aircraft communications control system (CCS) is a geographically distributed set of equipment designed to permit communication between individual crew members, between crew members and external facilities via radio links and to distribute audio signals from various sensors to the appropriate crew members.

In recent years, aircraft CCSs have been developed through a range of improved technologies. The early CCSs required that the receiver audio outputs and transmitter audio inputs were routed around the aircraft to serve each of the operators, see figure 1.1a. Gradually, the required number of operators and radios increased, leading to increased cable weight and perhaps more significantly to increased crosstalk between the wires which carry audio signals. These systems are known as distributed because all the electronic equipment associated with operators is distributed around the aircraft at the corresponding operator positions.

In order to reduce the crosstalk and weight of wiring significantly, systems were developed in which each operator and radio station was connected to a central unit via a pair of microphone / headphone leads and any required control leads, see figure 1.1b. These are known as centralized systems as they have most of the circuitry associated with operators located in the central unit. The main disadvantages with using centralized systems are that (a) each new system that significantly differs from the current product range requires a major redesign of the central units to meet the requirements of each new customer, (b) any major expansion also requires a redesign of the central units, and (c) unlike telephone exchanges, there are usually severe space and weight limitations in most aircraft.

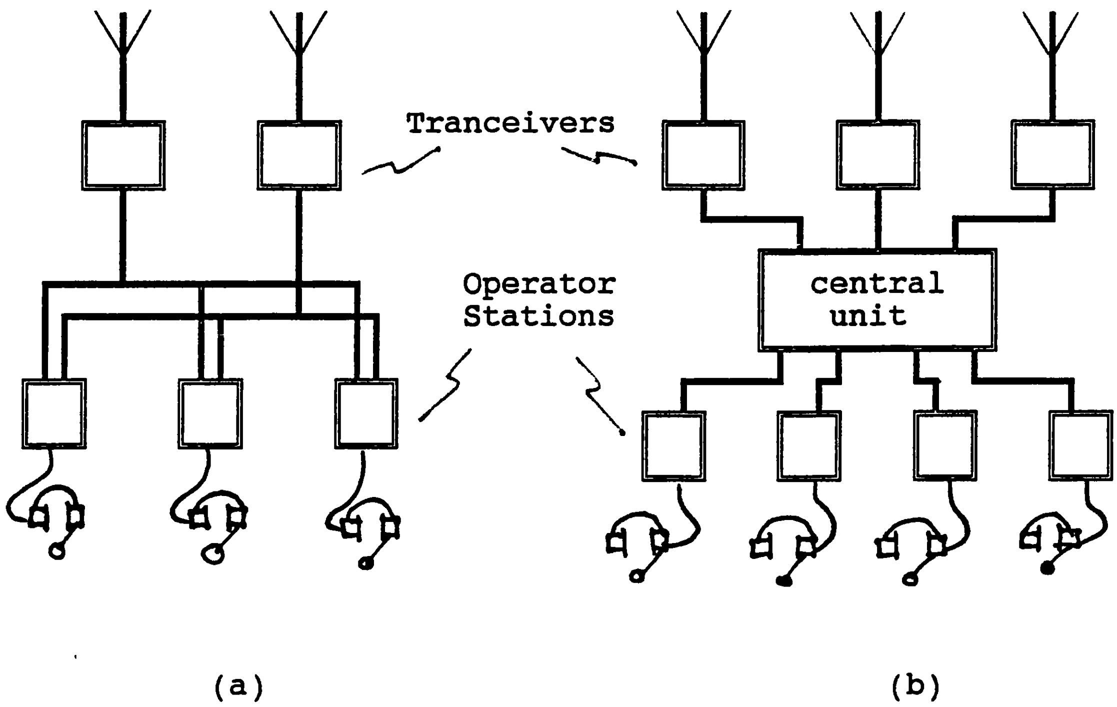


Figure 1.1. Aircraft CCS of recent years (a) distributed and (b) centralized

Two of the biggest advantages of distributed systems over centralized systems are (a) that they can be expanded with relative ease, and (b) although centralized systems can be designed to be very reliable by duplication of critical areas, in comparison with distributed systems, it is more straight forward to design a system with inherent integrity.

With increasing size and complexity of aircraft systems and increasing commercial pressures on development companies when designing new central units for customers, a new solution is required. The target is therefore to conceive a system which although distributed, does not suffer from the crosstalk problems associated with earlier systems. It is part of this thesis to show that by digitizing the audio and transmitting it along optical fibres on a local area network, these problems can be overcome. This is a new approach for aircraft CCS and should offer many improvements over conventional systems, and in all other respects should be at least equal in performance. The areas of desired improvement were intelligibility, crosstalk, signal-to-noise ratio, failure tolerance and radiation hardness [1].

In addition to distributing audio signals, a CCS provides the means for routing and volume control of audio signals received at a crew station. Therefore a basic requirement of the CCS is for reception and independent gain adjustment of several simultaneous signals for presentation at various operator stations.

In the early 1950's, the distributed systems described in chapter one were being manufactured for aircraft CCSs. The most recent example to come out of service was the Avro Vulcan bomber. Towards the mid 1960's however, a new generation of centralized systems were being conceived which could offer the operators far more facilities and connect a larger number of operators. Examples of aircraft that entered service during this period were the Wessex (distributed), Sea King (distributed), Hercules (distributed), Nimrod (centralized), Harrier (centralized), Phantom (centralized), BA Trident (centralized) and Gazelle (centralized). Then in the 1980's, several of the existing aircraft were upgraded with more modern systems until the present time when a new generation of aircraft CCS's is required.

Modern systems need to be more flexible, incorporating advanced functions such as data-link and satellite communications. They also need to be expandable and maintainable.

Flight safety has always been a key feature in aircraft CCSs. As a result of multiple failures or battle damage, the pilot and others, must be able to communicate with the outside world on emergency frequencies. This adds extra equipment to the system, thus departing from the original philosophy of minimising wiring. After a near total system failure, the flight essential features of the system must still function fully.

As the size of aircraft has increased, the length and bulk of cables has also increased so that they contribute significantly to weight, bulk, material cost and installation time and cost. Therefore to solve this problem some investigation must be made into the use of optical fibres and their associated advantages.

2. BACKGROUND

2.1 System Considerations

Using digitized voice signals on a distributed optical network has advantages in reliability, failure tolerance, expandability, crosstalk and TEMPEST. Some of the problems that will be encountered with a distributed digital voice system must now be considered.

As one objective is to replace a centralized system with a distributed one, some areas require special attention to take full advantage of a new system. Reliability is one important area that can be enhanced by specifying certain rules concerning the effects of failures and battle damage and is of such significance that it warrants a chapter to itself.

An important feature of a distributed system is that it can be made expandable quite easily as no central unit exists to restrict the maximum size of the system and therefore extra positions can be added by replication.

The range of system size that is intended to function must fit aircraft ranging from two seater for the smallest to twenty crew members for the largest. This however, is not the total number of connections to the network. For the aircraft with the largest number of communication facilities scheduled to be in service with the British Armed Forces this century, not only are there 19 operators but 14 radios, 20 receive only channels such as sonics and navigation aids, and ten encryption devices. To meet the requirements of the maximum number of types of aircraft, a network accommodating 80 nodes will allow for some expansion. While the preceding figures show the maximum size of a network required for aircraft use, a system will be more widely acceptable if it has the potential to be expanded to several hundred nodes either by increasing clock rates or by accepting the possibility of degradation of performance for increased size.

The above (along with application specific features), is going to govern bus capacity and must be considered with the size and complexity of each node as well as the size and complexity of the system.

2.2 Digitized Voice

The use of a real time digital audio bus will introduce several problems not previously encountered in aircraft CCSs. First consider the effect of data errors on digitized speech. These will cause distortion when the code is de-digitized as the final code will not be a true representation of the original code. One method of reducing the effect of data errors would be to incorporate error checking and possibly even error correction. A further cause of audio distortion will be any differences in delay between samples caused by the point-to-point path of a virtual audio link. This problem can be countered by appropriate buffering but if a packetized scheme is chosen, there will be a need to synchronise data either by time labelling or by some method of epoch definition (framing).

To fit in with the periodic nature of digitising and sampling voice signals there must be an upper limit on the transmission delay for voice data. This dictates a deterministic system rather than one with no upper bound on system access time.

2.2.1 Improvements with digitization

One improvement already mentioned, when digitizing audio, is that of crosstalk reduction. It also warrants re-iteration that the type of system proposed here would not be possible but for the improvements in crosstalk that can be obtained by digitizing the voice signals.

The delay concerned with the transmission of speech signals, on whatever medium (radio, co-ax etc.), will suffer because of finite propagation time. The acceptable level of this delay is governed by human factors. The human perception of delay falls into two categories; the delay between two speakers in a normal conversation and the delay between speaking and the speaker hearing what has been said. The time acceptable in the first case can be anything up to hundreds of milliseconds before the listener begins to notice long delays for replies and starts to interrupt the speaker. A good example of this is trans-Atlantic telephone conversations. A man in London talking to a man in New York via a land-line has no difficulty in conducting a normal conversation. This corresponds to a delay of less than 20 milliseconds. However, the same two men talking via a satellite link will have greater difficulty adapting to the corresponding delay of about 520 milli-seconds [3].

Delayed auditory feedback (DAF) (that is when a speaker hears his own voice delayed) or side-tone as it is more commonly known, has a tolerable delay which is significantly less than for the case described above. Laboratory tests have shown that test subjects find it increasingly more difficult to speak when the DAF approaches ten milliseconds. With this information in mind, a decision will have to be made later on whether to offer side-tone as a local facility where the microphone audio signal is passed into the headphones directly, or a global facility where microphone signals are passed throughout the distribution network before being presented at the speakers headphones.

2.2.2 Error Bursts

Although the distortion effect of data errors on speech has already been mentioned, it is also important to consider the effects of bursts of errors which may arise from the loss of groups of packets or momentary power failure or faulty switches or some re-routing process. Tests in the laboratory using standard test words and sentences (see appendix II) show that using 16 kbps CVSD as the coding scheme, by corrupting 256 sequential bits, an error burst of 16 milliseconds could be obtained. Repeating this error burst every half second seemed to make little difference to the intelligibility of speech in the system. If the number of error bursts is now doubled by cascading two error circuits, then 16 milliseconds of speech is lost every 1/4 second. The results from this experiment showed that test subjects could understand most sentences but only some words. The amount of continuous lost speech is 6.4%.

2.2.3 Dynamic Range

Research into the dynamic range of an aircraft CCS has been carried out by the Aeroplane and Armaments Experimental Establishment (A&AEE) at Boscombe Down.

A&AEE quote target figures for dynamic range of 60 dB nominal or 72 dB peak which on first reading seem to be much higher than expected. When the type of environment and the equipment used are considered, these figures look more realistic.

Some examples of background noise;

Oxford St (noon)	=	80 dB [3]
Quietest aircraft	=	85 dB (appendix II)
Inside a Nimrod	=	90 dB (appendix II)
Petrol lawn mower	=	100 dB [3]
Large helicopter	=	110 dB (appendix II)
Sea Harrier cockpit	=	135 dB *

* Worst case hovering at sea level requires greater than 5 dB isolation in helmet because pain threshold = 130 dB [3].

The required dynamic range of an audio system should be at least equal to the magnitude of the signal-to-noise ratio (SNR). Constituents adding to SNR in an aircraft CCS can be broken down as follows; the contribution from an operators helmet (6 dB) adds to the pain threshold as it provides isolation from the environment, deviations in level between different microphones can be as much as 6 dB. Operators are not consistent and may deviate when speaking by up to 6 dB and can detect signals buried in a signal-to-noise ratio of -6dB. After allowing for a difference of 12 dB nominal to peak, then adding the maximum pain minus background noise (45 dB), the worst case for required dynamic range is 71 dB.

A speech quality investigation will be necessary to evaluate what relative quality is required in the aircraft CCS and which digital speech coding techniques best suit the requirements.

2.3 Radio Management

In the past, aircraft radios have been selected and tuned in a variety of ways depending on operational scenario. Earlier systems had one control panel for each radio. Later systems had several panels controlling several radios (by radio numbers) by using the central unit to accept inputs from all panels then send outputs to appropriate radios. More recently, operators select a facility and the central unit manages the selection automatically by offering the operator a radio which suits his requirements for frequency, power, encryption etc.

The facility for automatic radio management was relatively simple to implement in a centralized system requiring however, a great deal of software and processing time to function successfully.

In a distributed system, with no central processor to oversee this function, successful operation will rely on a carefully conceived protocol.

The operational integrity of the system must be infallible, in that any single point failures or total unit failures will not "tie-up" or "lock-out" particular radio facilities. To this end, not only will virtual-path confirmation be required but also continuous checking such that if a failure does occur, then the radio facility can reset itself and be ready for another operator.

To combine digital voice and the facility for automatic radio management, the target for this project must be an integrated voice and control network.

3. EXISTING TECHNOLOGY

The topic being researched in this thesis encompasses a diversity of technologies. The study is for a system to fit into aircraft and therefore a knowledge of aircraft operation and environments is essential. Also, the area of digital voice coding must be researched to find the appropriate levels of performance. The more significant area of research in this thesis however, will be that of investigating the network distribution method.

It is intended to present in this chapter, a summary of relevant work that has been carried out by various institutions, to list some internationally recognised standards and show how that work has been used and continued to produce this thesis.

3.1 Aircraft CCS

Existing technology in and the history of aircraft CCSs has been covered in part in chapters one and two as an introduction for the reader to appreciate the target for this study. To recap, aircraft CCS have been developed and redesigned through progressing technologies for the past thirty years. A purpose of this thesis is to establish which technological directions, subsequent generations of aircraft CCS should take.

Much research has been carried out on aircraft CCSs by the main user of them, the Royal Air Force at their research establishments, two of them being the Royal Aircraft Establishment at Farnborough and the Aeroplane and Armaments Experimental Establishment at Boscombe Down, as well as at some commercial companies. Several papers have been published by these two sources concerning operational scenarios, CCS use, ergonomics etc. eg. Robertson [19].

To summarise the knowledge of aircraft CCSs available at the start of this thesis, there is a set of constraints involving bandwidth and dynamic range of the audio signals, reliability limits for equipment, and types of equipment that contribute to the formation of an aircraft CCS.

3.2 Digital Voice Coding

Research carried out in the field of digitizing coding speech signals is common and receives wide publicity because of its application to speech recognition systems and even more significantly, to private and public telephone networks.

Work on voice recognisers has been centred mainly around VOCODER techniques, which analyse the short term pitch and filter characteristics of speech. VOCODER research has undoubtedly led to a better understanding of the construction of the human voice, and has enabled subsequent researchers and engineers to develop coding methods which enable a significant reduction in the serial bit rate that needs to be transmitted so as to reproduce intelligible speech signals at the receiving end of the transmission path.

The alternative to vocoding techniques is waveform coding and several options are available. Details of some waveform coding techniques can be found in chapter 7. Established techniques for digitally coding speech exist, and it is not the intention of this thesis to present a new one.

3.3 Telephony

Work on telephone systems is based on point to point connections and the establishment of a virtual path between the two points. In assigning this virtual path, there exists a probability of unsuccessful connection.

3.4 Open Systems

Distribution networks for digital systems can in the first instance be broken into two categories; they are time division multiplex (TDM) and frequency division multiplex (FDM). The pros and cons of baseband and broadband transmission methods are discussed in chapter 4, but for the purpose of this chapter, only TDM methods will be considered.

3.4.1 Local Area Networks

The type of distribution network system being sought in this thesis falls into the category of local area network (LAN). A LAN is defined as a local digital network for bit-serial communication between independent devices and is distinguished from other types of data network in that it is moderate in size and that information is transmitted at a medium to high data rate on a physical channel [7].

The most important characteristic of a LAN is that it is confined to a private area, for example an aircraft. In most cases this area extends over less than 1 km. Up to several hundred data terminals communicate on a common transmission medium at data rates between 10 kbps and

100 Mbps. Using public networks and services or radio channels, LANs can be linked with each other over any distance.

3.4.2 Open Systems Interconnection layered model

Attempts at standardising communication protocols have been made internationally to reduce the incompatibilities of communication methods existing between data processing systems from different manufacturers.

Subcommittee ISO/TC97/SC16 of the International Organisation for Standardisation (ISO) created a seven-layer reference model for open systems interconnection (OSI), taking into account existing standards for data communication such as those generated by the International Telephone and Telegraph Consultative Committee (CCITT) in recommendations X.21, X.25 and HDLC (high-level data link control) [7].

In this layer model, information is transmitted by way of layer protocols between adjacent layers of a process, a communication therefore only taking place between the next lowest or the next highest layer of a system or between equivalent layers of different systems. Each of the layers of the reference model will be described in chapter 4.

3.5 Optical Fibres

Having decided to investigate optical fibres, it seems appropriate at this stage to cover a background of fibre optic work which applies to LANs. It is not intended to give a detailed discription of fibre optic operation. This will be found in [1,12].

3.5.1 Radiation hardness

It has been stated earlier that the area of application at which this thesis is aimed must be able to survive nuclear contamination (page 4). Although standard optical fibres darken with resulting high attenuation or blackout when subjected to ionising radiation, the newest fibres [9] are able to withstand neutrons, gamma and X-rays to reasonable limits in radiation hardening tests. By using these new radiation hard fibres, a survivable system can be conceived.

3.5.2 LED failure

The failure of fibre optic LED transmitters may cause problems of reliability in some types of LAN.

LED lifetime for even the most advanced fibre optic transmitters (Honeywell HFE4003) is quoted at 3000 hours when operating at full power. The use of these LEDs at low power (10% of maximum) for short distance point to point links will increase their mean lifetime to a figure of the order of 10^7 hours which enables the user to achieve a far greater system reliability figure. As, the chief failure mechanism is gradual rather than catastrophic, it is possible to use some form of monitoring to detect an unacceptable drop in LED output power.

3.5.3 TEMPEST

TEMPEST is the transmission or radiation of unintentional signals from an equipment, system or unit which upon reception and analysis may reveal compromising information.

The use of optical fibres will greatly reduce spurious transmissions making the detection of any secure information more difficult than in systems incorporating electrical distribution.

3.6 Standards

Some standards have been released that are of interest to this thesis. The most relevant ones are listed as follows:-

3.6.1 British Telecom

Research into using a digital solution for telephony has been performed mainly for British Telecom (BT) in the UK and standards are available, having been generated by the CCITT. The main examples of relavant CCITT standards being 30-channel PCM [4] in the UK and 24-channel PCM in the USA.

A 30-channel PCM multiplex system [5] provides a transmission capability by digital means of 30 telephony circuits. Each system consists of terminal multiplex equipment and an associated digital-line system.

In the frame structure of 30-channel PCM, each audio channel (4kHz) is sampled at a rate of 8 kHz (every 125μs). In each of the 125μs periods is contained an amplitude sample of each of the 30-channels and this period is known as the frame. Each frame is divided into 32 time slots and these are designated TS0 to TS31; the two extra time slots are used to convey frame alignment and signalling information. Each amplitude sample contains eight binary digits. The gross digit rate can be obtained from the frame repetition rate, the number of time slots and the number of bits per sample.

Gross digit rate = 8000 x 32 x 8 = 2.048 Mbps

Although 30-channel PCM is unsuitable for the requirements outlined in earlier chapters, this suite of CCITT standards [4] also contains information on jitter and baseband transmissions which will be of some use later.

3.6.2 Orwell

Research carried out by Adams and Falconer [6] at British Telecom Research Laboratories is more relevant. Their solution to a partially related problem is called the Orwell Ring and uses a 34 Mbps slotted ring. The Orwell Ring will be discussed in more detail in chapter 5.

3.6.3 IEEE 802 and FDDI

The LAN committee (project 802) of the Institute of Electrical and Electronic Engineers (IEEE) have standardised on three different types of LAN corresponding to the first layer of the OSI reference model. The American National Standards Institute (ANSI) have taken one of the IEEE 802 standards further to produce the Fibre Distributed Data Interface (FDDI).

The standards correspond to;

802.3 = Carrier sense multiple access with collision detection
802.4 = Token passing bus
802.5 = Token passing ring
and the ANSI standard
FDDI = Token passing ring

These four standards will be discussed in detail in chapter 5.

3.6.4 Overheads

The LAN standards offered by IEEE 802 and ANSI FDDI have one serious drawback which warrants a mention at this early stage. To cope with a diversity of applications, the IEEE have catered for most situations by building versatility into their packet header structure. Figure 3.1 shows an example of the supernumerary bytes in a typical IEEE 802 scheme.

The abbreviations used for figure 3.1 are:

SD = Starting delimiter
AC = Access control
FC = Frame control
DA = Destination address
SA = Source address
FCS = Frame check sequence
ED = Ending delimiter
FS = Frame status

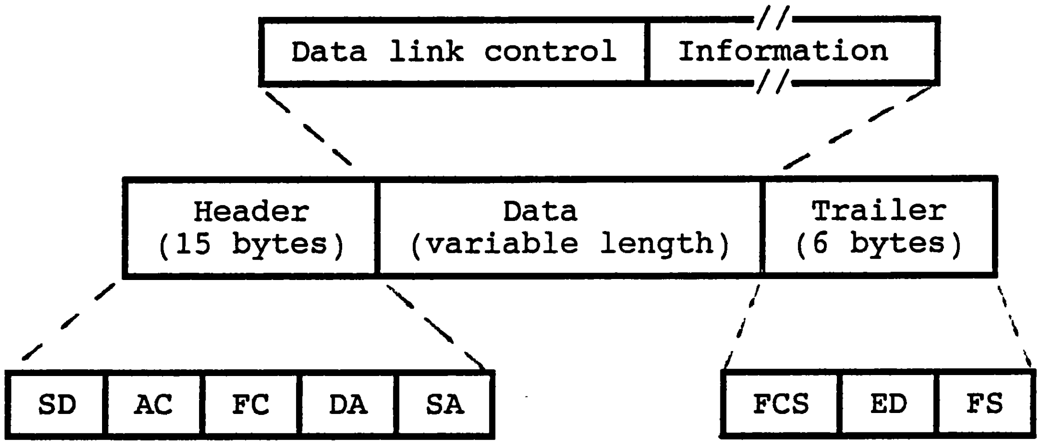


Figure 3.1. Packet structure for IEEE 802.5

It will be shown in chapter 5 that a short header is desirable to avoid making the network operation too inefficient.

3.6.5 STANAG 3910

During the period of research leading to this thesis, a new aircraft bus standard was released by NATO, draft STANAG 3910 [8] and therefore warranted investigation.

The STANAG 3910 and subsequent EFA-BUS [8] standards, although lacking in size and expansion capacity, has some fibre optic features which are of relevance to this thesis.

Assuming certain compatibilities with STANAG 3910 and EFA-bus [8], it will be worth trying to adopt features which are laid down by those responsible for standards, recognising that these may be market driven.

4. MODEL DESIGN

Although the idea of developing a local area network (LAN) to include coded speech is not a new one, the particular problems involved in developing an intra-aircraft audio distribution network are different to those encountered in previous studies of LANs and to those encountered in telephone networks [6,15].

The operational philosophy being conceived for an aircraft with varying mission roles dictates that an initial system model must comply with certain constraints.

All variations of system utilisation from the minimum to the maximum constraints being equally probable means that the system capacity for this particular networked system must be viewed in a different way to others.

4.1 Information Transfer

The two types of information that must be transferred through the networked system are for voice and control. There are major differences in the requirements for transmitting these two types of data and therefore a model which accommodates both must be conceived. This model will be a compromise for the two types of data.

4.1.1 Voice Information

The real time nature of voice communications requires that voice information be delivered with small transmission delay and in near synchronous fashion. Conversely, it is relatively tolerant to data errors.

The communications control system of a mission based aircraft is such that the normal practice of not transmitting quiet periods of digitised speech is not practical. This is because aircraft operators are trained and experienced in detecting audio signals buried in high levels of noise. It therefore cannot be the responsibility of the audio system to reject any signal that may be valid. To this end, all audio transmission will be initiated by the activation of a "press to talk" key. Transmission will occur as long as this key is active.

The normal operational complement of an aircraft can vary from just two operators conversing on an intercom channel to all operators using all radios and several sensor inputs. In both cases degradation and "un-made" connections are undesirable.

4.1.2 Control Information

Control information, unlike audio information is not sensitive to delay. Within reasonable bounds, control signals can be queued for several milliseconds (the limit being the acceptable response time when an operator uses his control panel) and then an altered sequence can usually be tolerated. Data errors however, are totally unacceptable. If an integrated service is to be provided for transmitting both audio and control information, it must either have error detection and correction for the control information or redundant error handling capacity for the audio information. A reasonable solution is to give the control information a lower access priority in any network connection. This eliminates any question of increasing audio delay and will help to maintain an upper bound on any such delay. Errors in control information can be countered by use of parity and a continuous check for response such that retransmission is available on request.

4.2 Starting Point

The starting point for the model is covered in this section. Topics that will affect the model are discussed, but areas such as bit rate and size should be transparent to the model.

Several aspects have to be considered before an accurate model can be constructed. One object is to conceive a system that will use few types of different boxes (ie. aiming for commonality). Apart from the obvious economic advantages (which are not necessarily relevant to a model design), improvements in manufacturing, maintainability, customer storage and training should be made. As stated in chapter 2, one requirement is for expandability and therefore variable system size. Next we must discuss some basic considerations.

4.2.1 Questions

Certain questions arise when trying to replace conventional systems with a real time, digital audio bus. Firstly, although simultaneous reception of multiple conversations is possible, only one speaker can be understood by a listener at any time. Therefore, whether multiple channels will be offered to operators must be decided. With all channels offered, the operator will be able to decide which channels he wishes to listen to. To make the job of channel rejection unnecessary, it is desirable that the maximum number of channels that can be listened to, is equal to the number of channels that can be selected on each operators control panel. In other words the CCS should not decide what is suitable for rejection.

A further question arising from simultaneous channel reception is what effect it has on buffer storage size at each operator node and the effect on any signal processing algorithm.

4.2.2 Medium Access

An initial choice exists between broadband and baseband techniques for data transmission.

Time division Multiplex (TDM) and Frequency division Multiplex (FDM) systems have both been shown to adequately perform various functions [16,18]. To obtain an adequate bandwidth in an FDM system for a number of carrier frequencies, a co-axial cable system would have to be used. Careful design of terminals would be needed to avoid creating standing wave patterns on a passive bus, or a point-to-point architecture could be used. In the latter case the quantity and complexity of wideband RF circuitry would be increased, together with the size of the terminal. Retrieval of a modulated carrier from the bus could be accomplished by using a single conversion superhet type of circuit, with the local oscillator frequency set by a synthesiser. The high frequency IF circuits could be eliminated by using homodyne, or zero frequency IF techniques. This would, however, probably generate a requirement for a wideband RF buffer amplifier to avoid corrupting the carrier on the bus with local oscillator leakage through the mixer.

The number of nodes that may be connected to a LAN for use in an aircraft CCS would dictate a FDM system requiring carrier frequencies in the megahertz range. This will make mandatory, the use of LC selective circuits, which tend to be bulky, and high quality screening to reduce EMC problems in associated low frequency circuits to tolerable levels. This will lead to terminals which are significantly greater in volume than those required for digital TDM systems. This would make the systems less acceptable to potential customers. The volume problem would be considerably exacerbated by the requirement to extract the modulated information simultaneously from a number of carriers. The increase in volume is because each terminal with a multichannel simultaneous receive requirement would need to be fitted with paralleled receivers and a synthesis system capable of generating several unrelated frequencies simultaneously. In order to achieve any degree of miniaturisation of a FDM design it would, in general, be necessary to raise significantly the frequencies of the carriers in order to obtain smaller component values and sizes. An action of this nature brings further problems eg. effective filtering. In addition, isolating and screening are more difficult to achieve and components tend to be more expensive.

If amplitude modulation of the carriers is used, the required degree of screening would approach that normal for radio equipment, adding to weight and bulk. Frequency modulation should be somewhat better in this respect, since direct demodulation of the signal by audio circuits is unlikely. FM leakage could, however, cause distortion effects in other circuits by induced saturation effects.

The conclusion is that a FDM based bus system is not suitable. The operational and size constraints mentioned above are decisive in making the choice to adopt a TDM rather than a FDM system for aircraft CCS purposes.

The use of TDM means that optical fibres may be adopted. FDM would have excluded their use as multi-chromatic switches are not achievable with present technology.

4.2.3 Network and System Control

There are two basic groups into which types of network control may fall; centralized and distributed. Each has advantages and disadvantages depending on the application and environment in which they are being used.

For an aircraft CCS, reliability is of paramount importance in that there must always be a method of communication between crew members. To maintain reliability using centralized control, the controller and bus must be at least dual redundant, with monitoring and supernumerary bits so that control can be switched to a second bus controller if necessary. With distributed control, redundancy can be inherent. If one terminal fails, then it can be either switched out, or an alternative bus switched in. In this case, a dual bus can still be used when the distributed network becomes severed.

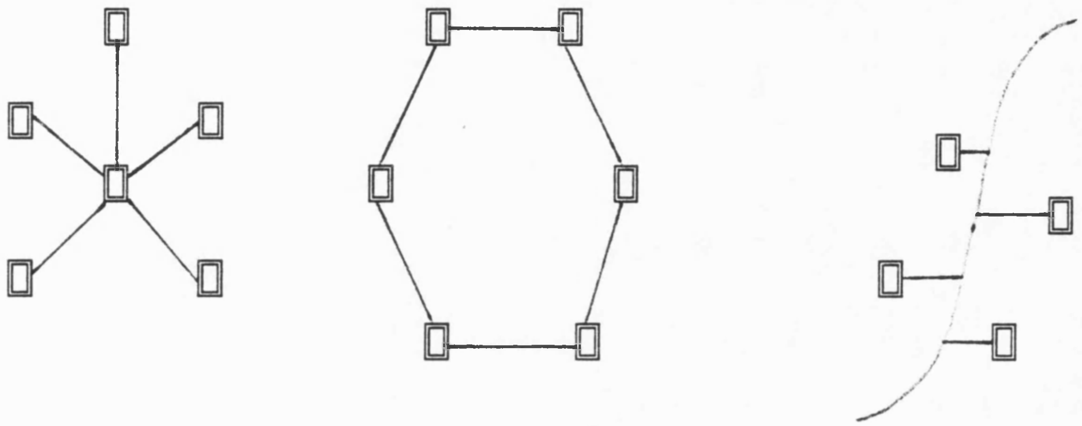
The fact that centralized bus control requires an extra control unit per redundant bus means that extra circuitry will also be required at each terminal. Distributed control makes each terminal more complex and inevitably larger due to the absorption of the functions originally performed in the bus controller. However, with the use of modern gate array techniques, this can be achieved without a substantial increase in circuit size.

Software used in a centralised bus control implementation is replaced by a carefully designed protocol when utilising distributed control. This has the advantage that the new area of complexity in a distributed system is duplicated to obtain the equivalent level of performance to that of a centralized system and the complexity now occurs much earlier in the design process.

The advantages of a centralized control strategy are detailed in [17], but for this application it has been shown in the introductory chapters that a distributed control philosophy will be most suitable.

4.2.4 TOPOLOGY

Types of LAN fall into three basic categories, they are the star, the bus and the ring (figure 4.1). The active star architecture was the method used for audio distribution systems in the past. These star shaped systems have the disadvantages of being susceptible to single point failures and requiring re-design for each new system. A passive star may be used, but this has problems of attenuation when using a fibre optic coupler as the centre of the star, the result being to limit the maximum number of nodes to around 16 [8]. The bus and ring architecture however, lend themselves to a degree of distribution and versatility and therefore merit further investigation.



STAR RING BUS

Figure 4.1. Network Topologies

For the purpose of this model, whether a ring or a bus topology are adopted is transparent. This will become appareant in section 3 of this chapter.

4.3 OSI Reference Model

The Open Systems Interconnection (OSI) model of BS6568 [13] has already been introduced in section 3.4.2 and will now be described in more detail. It can be used as a tool to decompose the concept of an advanced CCS into a layered and adaptable design where each of the layers in the design communicate with adjacent layers and no other part of the design.

By using this layered technique in the construction of a model, it is possible to arrive at a design which is flexible enough to accommodate changes of requirements, specification, protocol and configuration at any level without affecting other layers in the design, as long as the communication interface between layers is maintained.

The OSI reference model contains seven layers. The seven layers comprise four transport-orientated blocks (layers 1 to 4) which are normally specific to a type of LAN and three application-orientated blocks (layers 5 to 7) which always depend on the system and its requirements. To permit communication between the users of a network, layers 1 to 4 must be implemented. They are mainly concerned with problems of information transport. The subsequent layers 5 to 7 require the co-operation of the participants in communication.

The following individual functions of each layer are defined:

1. The physical layer comprises, as a basic condition for communication, the hardware such as the medium and the interface characteristics.
2. The data-link layer provides for addressing of the information to be transmitted and ensures that transmission errors occurring in the physical layer are detected and corrected.
3. The network layer ensures optimum routing of information through the network. Here different requirements such as security and speed of data transfer can be taken into account.
4. The transport layer provides for setting up and clearing down end-to-end communications, as well as for the required flow control and error detection.
5. The session layer manages the information exchange between users. This task also includes the decision as to the form in which information exchange is to take place.
6. The presentation layer processes the transmitted information such that it can be optimally used by the participants in communication.

7. The application layer is the top layer and constitutes the interface to the user, its standardisation depending on the user's requirements.

The OSI layered model is the basis upon which further work can be carried out; the implementation of various functions being left to circuit designers and software writers. It can be seen that the layered reference model was conceived predominantly for computer manufacturers, but even so can lend itself to manipulation to any kind of networked open system.

4.3.1 System Description

The diagram in figure 4.2 was developed as a novel visual aid. This figure shows that a layered approach can be used by following a path through the model in a clockwise direction to achieve a practical solution. Starting with the OSI reference model at the top of the diagram, the requirements for a future system can be decomposed to form a set of system related objects that can compare directly to the seven layers of the starting point.

The objects in the system section of the diagram relate to independent communication channels and can be considered separate to all other layers in the model.

They correspond to a very basic set of requirements which are merely to distribute information between users (operators and radios) and at the system stage each object is an empty box waiting to have a function associated with it.

The system aspects of the model can be further decomposed into functional blocks that are required to meet the system's objectives. In the process of decomposing system objects into functional blocks, it must be remembered to isolate by means of a communication path, those functions in different layers. Should detailed requirements change, it would then be possible to accommodate either a change in audio coding scheme with no effect on the distribution or routing mechanisms or, a change in medium access processing with no affect on the communication interface that each other layer sees.

Finally, as will be shown later in this thesis, a practical solution will follow from the functional description. Note that solvable blocks can overlap layers and perform more than one function. This is because it is the interfaces between the layers that are important and therefore an interface between two processes may exist in one processor.

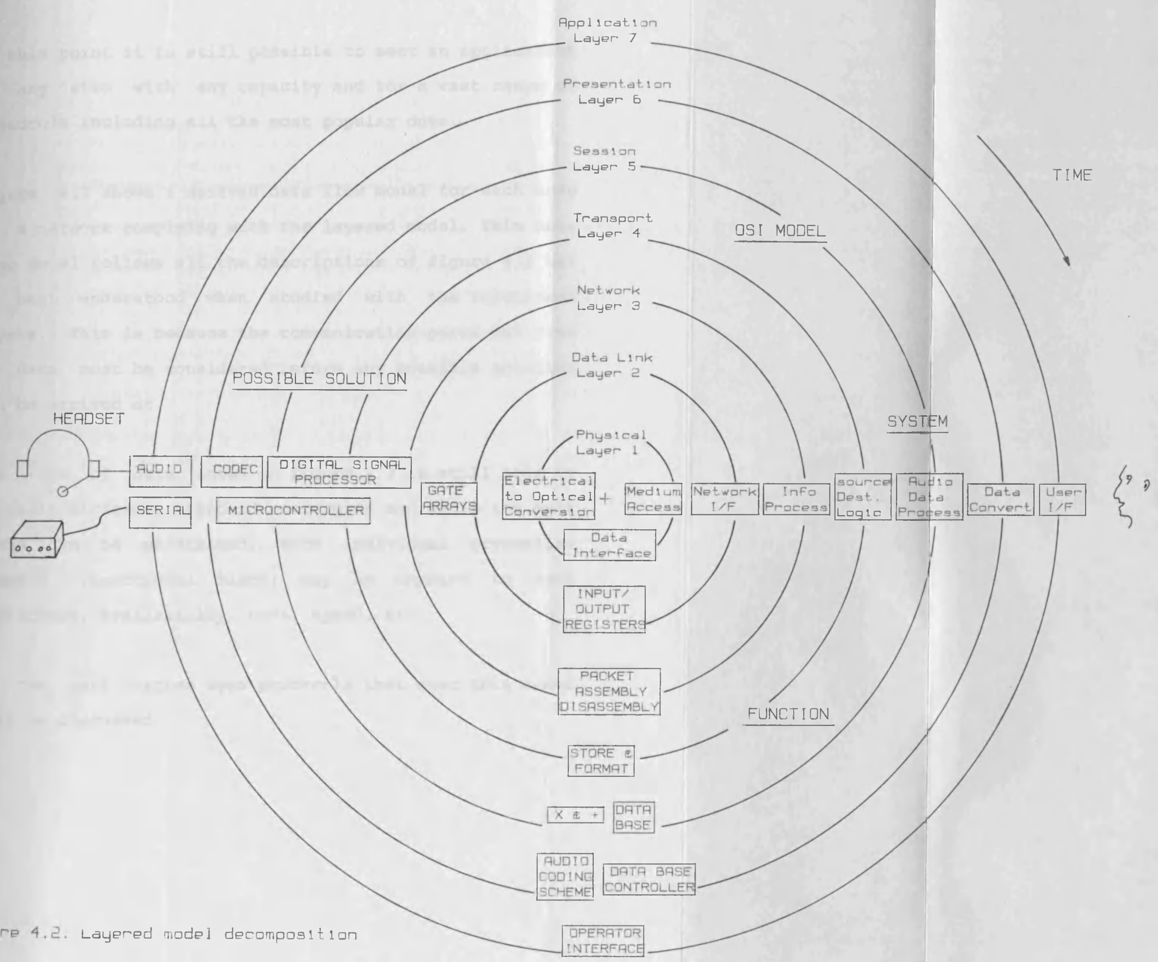


Figure 4.2. Layered model decomposition

At this point it is still possible to meet an application of any size with any capacity and for a vast range of protocols including all the most popular ones.

Figure 4.3 shows a derived data flow model for each node of a network complying with the layered model. This data flow model follows all the descriptions of figure 4.2 but is best understood when studied with the functional layers. This is because the communication paths and flow of data must be considered before any possible solution can be arrived at.

The flow of data shown in figure 4.3 is still able to maintain different standard procedures and while the data paths can be maintained, each individual processing element (functional block) may be changed to suit preference, availability, cost, speed, etc.

In the next chapter some protocols that meet this model will be discussed.

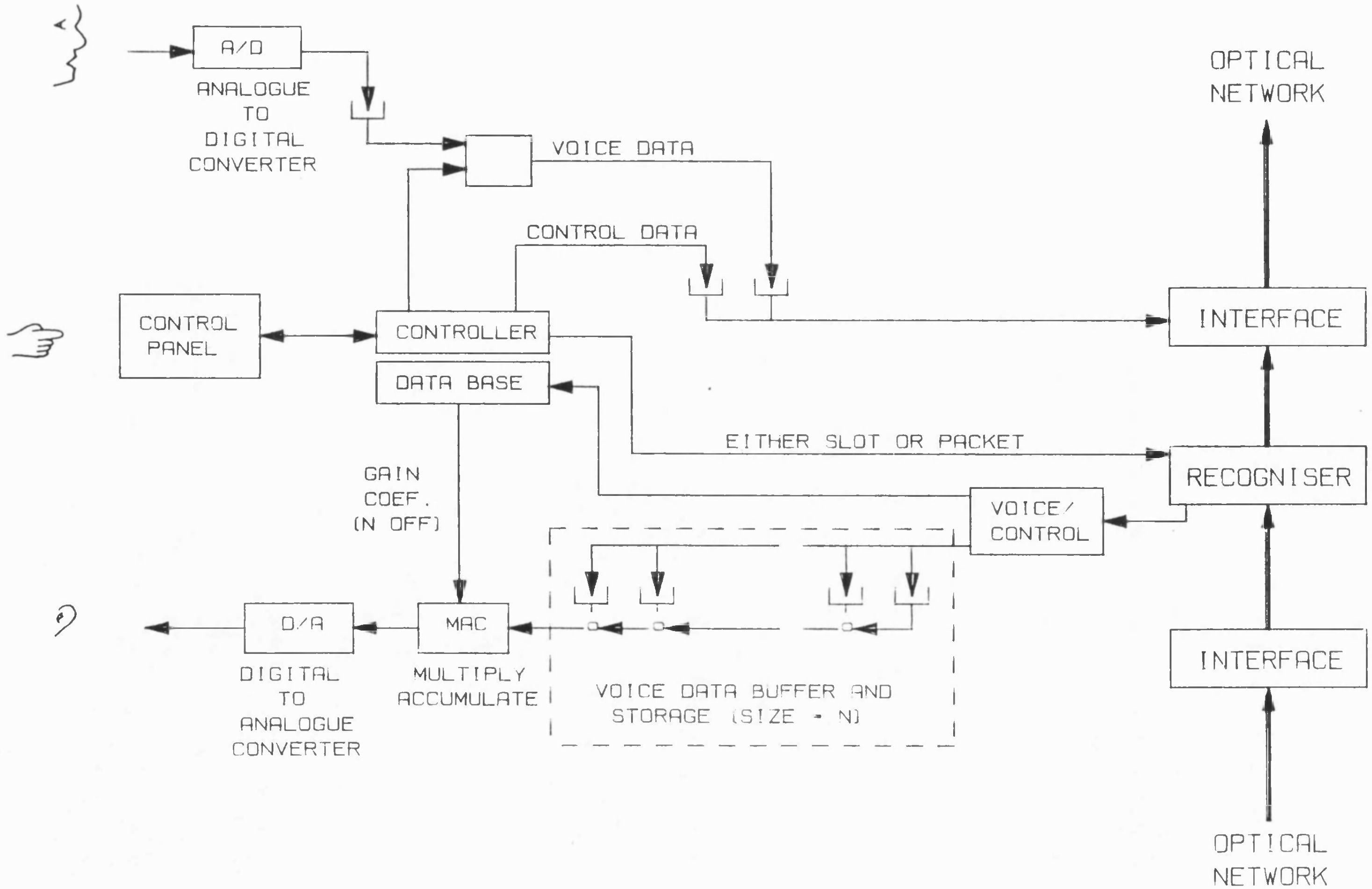


FIGURE 4.3. NODAL DATA FLOW MODEL

5. LOCAL AREA NETWORKS

The requirements which must be met by a protocol to be developed for this LAN are that the system should be easily expandable without significant redesign, rewiring or degradation of performance. The system must accommodate a heavy workload ie. several nodes transmitting continuously. It must support simultaneous reception of multiple signals and be able to integrate audio and control signals. Finally, the failure handling of the system must be such that single point failures cause no degradation and multiple failures cause only graceful degradation.

The example of 30-channel PCM already outlined in chapter 3 (ISDN) fails to meet the above requirements for two reasons; Firstly, the system is not easily expandable because of the 32-slot limit and secondly, integration of voice and control data is more difficult than for packet switched networks because slots have no address and therefore each slot is dedicated to one channel, at the time period required for voice reproduction.

5.1 Packets

Packet switching is a common feature in most LANs, where the data is transmitted as a packet containing necessary housekeeping including synchronisation bits, destination and source address bits, user data and check digits. LANs have a fixed address range dependent on implementation ranging from 8 bits (256 nodes) to 48 bits. The latter is intended either to be used in a structured way, ie. broken down into components such as organisation, site, network and host, or to provide a unique address for every machine worldwide. Depending on the network, the data size may vary considerably, the range being typically between 2 and 2000 kbytes. Generally, the whole packet is checked using parity or a cyclic redundancy check. The housekeeping information is very specific to the topology and control technique used.

Packets on local area networks generally consist of the following [18]:

- synchronisation
- control
- source address
- destination address
- data
- error check.

A packet consisting of the above is ideally suited to the transmission of digitised voice signals around a networked system on the condition that one works within certain constraints. It will be preferable to keep packets short because of the delay sensitive nature of speech data. However, it is still possible to integrate large amounts of data successfully by cascading multiple short packets to effectively provide one large packet. This is only possible if access to the network by short voice packets is not denied.

5.2 Bus Protocols

To meet the requirements listed in previous chapters, only those methods that distribute control throughout the network will be considered here. Polled techniques such as MIL-STD-1553 have already been discounted on the basis that they use centralized control. There are two basic categories of distributed bus protocols; they are contentious transmission and managed transmission.

5.2.1 Contention Buses

Contention based buses broadcast onto their transmission medium in the same way that radios transmit into the air-waves. In fact, there is no reason, other than privacy and fear of interference, to prevent the use of radio broadcast for local area networks if this is suitable for the application.

Probably the first of the broadcast packet networks was the ALOHA system. Although it is of mainly historical value now, it has the virtue of simplicity and can throw some light on the way current schemes have evolved. The University of Hawaii computer centre based in Honolulu serves a large number of terminals scattered around many of the islands in the Hawaiian group. To provide an easy means of access from these terminals, a radio broadcast medium was used. Naturally this medium is accessible to everyone with the right equipment and therefore if more than one node attempts to transmit at the same time, the airways will carry two packets at the same time, and therefore both packets will be corrupted.

A progression of work on the ALOHA system led to perhaps the best known LAN work to date, which is the development of Ethernet at the Xerox Palo Alto Research Center. Xerox re-specified their bus (a 10 Mbps network) in the late 1970s as a proprietary standard in collaboration with DEC and Intel. In 1985 the Institute of Electrical and Electronic Engineers standardised this network as the Carrier Sense Multiple Access with Collision Detection (CSMA/CD), IEEE 802.3 [36].

This bus protocol normally uses co-axial cable as its transmission medium and works by allowing any piece of equipment connected to the network to transmit onto the transfer medium as long as no other terminal is already transmitting.

This technique has the disadvantage that if two terminals try to transmit at the same time, a collision of data will occur causing the data packets to be corrupted. Once the collision is detected, each terminal then waits for a random delay before attempting to retransmit, ensuring that the same two terminals do not collide a second time. However, successive collisions can occur with other terminals.

Transmitting speech data on CSMA/CD has two serious drawbacks. Firstly, the algorithm has no upper bound on access delay, and secondly, although data can be successfully transmitted at higher bit rates, the time for collision detection remains constant for a given network medium and therefore becomes increasingly more significant.

5.2.2 Token Passing Bus

The alternative bus protocol is the token passing bus (standardised to IEEE 802.4). Its token access scheme has a special bit pattern called the token which is sent out onto the network. When a node wishes to transmit, it must first be given the token by another node. The transmitting node then holds onto the token until it has finished sending its data, and then sends the token out as the last part of the information.

This scheme has a complicated management protocol compared to its ring counterpart and the token passing activity can be shown to consume a significant proportion of the available transmission time. Whilst token passing is conceptually simple, in practice it can be complex to implement because of the need to provide specific mechanisms for start-up, to overcome loss of a token, to prevent duplicate tokens and to allow for a node to enter and leave the network. When a ring network is used, the access management problems become considerably simpler and at higher bit rates the ring becomes more attractive and with its repeater station, lends itself to use with optical fibres, making paths between terminals more reliable and fault tolerant.

5.3 Ring Protocols

The ring architecture can be realised in three basic forms, they are the register insertion ring, the slotted ring and the token passing ring.

5.3.1 Register Insertion

The register insertion ring operates by allowing the node which wishes to transmit to insert a register into the path of the ring. The register is the length of a packet (that is, all the information which the node wishes to transmit in one go) and is withdrawn when the packet has completed a full cycle of the ring.

Provided the register insertion is carried out when no data is flowing around the ring or when the end of another packet has passed there will be no confusion. The result is that the node's data is shifted out of the register, around the ring, and other data passes through the register in the meantime. At the receiver a second register is used into which all data passes. At the appropriate moment an address check is made and if the destination in the packet matches the node address, the packet is copied (parallel) into another register. When the original packet has made a full circuit the control logic checks the source address against its own and on a match removes the register from the ring.

It will be noted that the packet does a full circuit of the ring. This provides a passage for low level acknow-ledgement information between recipient and sender, eg. to indicate busy or accepted.

The only way a sender can recognise it's transmitted packet is by it's source address. If this gets corrupted in transit the node would become deadlocked. Additionally, the "zombie" packet would circulate indefinitely. This problem is solved by having a time-out in the node, which resets the node and removes the register from the ring. Additionally a monitor node is provided in the ring to remove zombies.

In addition to recovering zombies, the monitor can perform initialisation at power-up, ensuring that the ring is empty. Random circulating bits could give rise to spurious reception.

5.3.2 Slotted Ring

The slotted ring operates by making use of the delay that is inherent in a large ring and splits it into a number of inter-slot gaps and slots for inserting packets. A 10 Mbps standard exists for this system in BS6531 [40]. This is based on the Cambridge Ring.

The inherent delay in a typical 10 Mbps configuration consists of cable delays of about 4.5 ns/metre, ie. 100m of cable delays a signal by 450 ns, this can be thought of as storing 4.5 bits. Repeaters and switches add at least a further 2.5 bits delay, so that a 400m ring with 10 nodes stores about 43 bits or with 80 nodes stores 218 bits and can be thought of as a circulating shift register.

Early in each slot is a full/empty bit. Any node wishing to transmit, can, on seeing an empty slot pass through, mark it full, and fill the remainder of the slot with a mini-packet.

Like the register insertion method, the slotted ring uses a monitor which plays several management and maintainance roles. For very short rings there may be insufficient storage to hold even one slot, so the monitor has a shift register which can be added in series in such cases.

By virtue of its principles, both cable and repeaters must remain in circuit even when the host computers attached to the nodes are powered down, thus, power for the repeater section of each node is distributed around the ring itself. In practice several power supplies are provided to allow for power supply failures.

An examination of this protocol suggests it is not suitable for an aircraft system as it stands, because firstly, the bit rate is insufficient to accommodate the required bandwidth, secondly, the delays achieved using a small optical ring are too small and variable to maintain a reasonable slot size, and thirdly, the central role that the monitor plays makes the network less distributed and susceptible to single point failures.

The characteristics of both the register insertion ring and the Cambridge ring are similar. Both offer very low latency transmission, the allocation of bandwidth provides equal access opertunity between nodes. They are ideal for switching synchronous data streams such as voice because of their small packet size and predictable behaviour. Using these simple protocols a larger packet can be built allowing their use in other applications.

The small mini-packet size allows low-level flow control and simple hardware interfacing but does waste about 50% of the bandwidth.

5.3.3 Token Ring

The token passing ring in steady state operation continuously circulates a token, usually consisting of one byte. When a terminal has a packet to transmit, it extracts the token from the ring, transmits its packet, then finally retransmits the token.

In principle, once the token is held, very long packets could be transmitted. Normally, a limit is set to how long a token may be held, so that latency is held within limits and so that a lost token failure can be quickly detected. In some token rings a node may only transmit one packet while it holds the token, in others several packets can be transmitted.

One advantage of the token passing ring is that each time a node passes the token on, the down stream node will be the first to be offered the free token. In this way, every node gets the chance to take the token.

A disadvantage of token passing rings is that the supernumerary bits involved in overheads and management are quite large. This is of little significance when dealing with large packets for computer data but for the small packets required for voice data, this system becomes very inefficient. Two standards are available for token passing rings, they are the 4 Mbps IEEE 802.5 [38] and the ANSI 100 Mbps Fibre Distributed Data Interface (FDDI) [39].

5.4 Network Coupling

With LANs that are confined to a small area or private premises, coupling to an adjacent or remote LAN is very often necessary. This requires coupling elements, which can roughly be divided into two groups.

BRIDGES are used to connect two identical networks. Because they are the same, no protocol conversion is necessary between the networks and the access methods are identical. A bridge is only required to recognise whether information is addressed to its own or the other network. Bridges handle layers 1 to 3 of the OSI reference model.

GATEWAYS connect two different types of network. Depending on the degree of difference, gateways have to solve problems of varying complexity such as data-format modification, protocol conversion, transfer-rate conversion or handling of different access methods. If the application functions of the two networks are compatible, the gateway is only required to convert the transport functions (layers 1 to 4). However, gateways including application conversion (layers 1 to 7) must be used when systems are entirely incompatible.

5.5 Summary

Existing standards could possibly be used to meet aircraft communication system requirements with varying degrees of modification. However, the process of modification renders these methods non-standard and therefore the reasons for using them disappear. The obvious choice now, as a non-standard must be used, is to research the optimum non-standard.

Much work has been done on the efficiency of standard protocols [50-52]. In the worst case for a contention bus, Davies et al [52] quote a figure of approximately 18% efficiency for bandwidth utilization of a pure ALOHA system. Use is made in this thesis of the relative efficiency analysis from the relevant references.

The derived protocol is necessarily a hybrid of existing protocols taking those features of each which are best suited to operation with voice and for use within an aircraft system.

The resultant LAN is a practical register/slotted ring which can make use of a full/empty management technique. The inserted register part of the network is physically realisable because of the relatively small packets that are required of a voice centered LAN, the main criterion now being to minimise the quantity of overheads and supernumerary bits so that they do not make a significant contribution to the overall packet size and delay.

6. THE PROPOSED RING PROTOCOL

This chapter details how information can be transferred around a ring network that is used to communicate between adjacent units in the Digital Communication Control System (DCCS). The ring forms a network which is an essential part of a high integrity system for operation in an aircraft environment. Word formats and packet structures are also discussed.

In the sponsoring company, the proposed LAN has become known as the Russell Ring and therefore will, at times be referred to by that name.

Having established that a ring network is to be used because the two types of distributed bus access are unsuitable, and that optical fibres will be incorporated to take the fullest advantage of the point-to-point connections within a ring, those features of slotted, token passing and register insertion rings that best suit the requirements (eg. bounded latency and ordered management) will be discussed.

6.1 Requirements

The first and most significant requirement to have an effect on the digital voice system is the need for each operator station to interpret several audio signals simultaneously. eg. listen to audio sensor inputs, radios and intercom at the same time. This requirement may vary between 1 and about 20 (the Maritime Reconnaissance Nimrod can have 19 channels selected for reception at some operator positions). The simultaneous samples resulting from multiple sources must be decoded and processed within the same time frame. If the decoding of samples were not carried out in ordered and sequential time frames (epochs), then the reproduced audio signals would be distorted. This is clearly unacceptable. Several options for maintaining epochs are available. The idea of one node generating frame boundaries would work, but this would introduce a centralized element, thus making the system less reliable and vulnerable to single point failures. A further method of accomplishing clock, and therefore frame generation from one node at a time, is by using a token based system. In figure 6.1 node B has the token and provides the master clock. When node B has completed its transmission, the token will be offered to other nodes. The clock generation will then be taken on by the new node with the token, as this is the only node permitted to transmit.

An equally acceptable option is a real-time label on each packet, with each node having an on-board clock such that a field containing the absolute time can be included with transmitted samples. The ability exists in each node however, to generate its own local epoch based on a multiple of its sample period. If this is a derivative of the reconstituted data ring clock, the epoch will have identical duration if not coinciding with boundaries in other nodes.

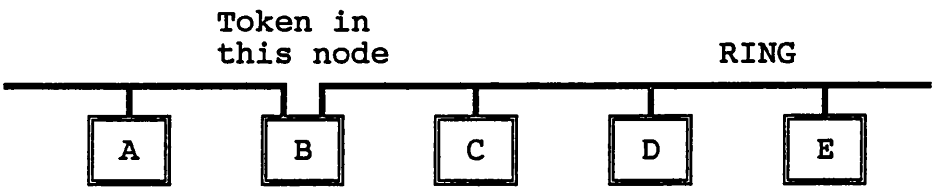


Figure 6.1. Clock source in token-passing rings.

A locally generated epoch also guarantees that side-tone, if permitted to travel the entire network, will be decoded one epoch later, because the ring transmission delay will be less than that acceptable for side-tone (side-tone delay dictating the duration of speech transmitted). A final decision on whether to decode side-tone after a complete cycle of the ring, can only follow when a ring protocol has been established.

Further requirements which will have an effect on the conception of the final network are:

expandability

open microphone inputs (nodes transmitting continuously)

single failure tolerance and

minimal degradation caused by double failures (graceful degradation)

All features of the system, including all aspects from ring medium to the operator interfaces, will be connected and communicate in a layered manner. For hardware connections, data-flow must be carefully considered to meet timing requirements. For software, the passing of data variables between layers must be formally specified to ensure that it meets system requirements. Thus single layers will be interchangeable with other designs should a requirement arise at some later date.

6.2 Some Reasons for Directions

Some observations still must be made before an optimum network may be derived. Creating a new LAN, starts with several options. A different choice of direction in the early stages may result in a completely different solution. As it is the optimum solution that is sought in this thesis, those topics which affect the optimum solution will be discussed in this section.

6.2.1 Packet Data Size

As discussed in chapter 4, a major difference between speech transmission and data transmission is that data is error critical and speech is delay critical. To this end, the elapsed time of transmitted speech in packet form will represent the largest portion of any delayed auditory feedback. This means that packet length must be kept relatively small.

The required length of each packet can be derived from a combination of certain system parameters. If the elapsed time for storing speech prior to transmission is D, the packet length in bits is L_p, the number of samples in each packet is S, the maximum number of nodes connected to the network is N, the network bit rate is R and the sample rate is F, then the network bit rate can be defined as;

$$R = F\ N\ L_p\ /\ S \qquad (1)$$

when related to the sample rate, or

$$R = N\ L_p\ /\ D \qquad (2)$$

when related to the maximum number of nodes. (1) and (2) reduce to;

$$S = F\ D \qquad (3).$$

This shows that the number of samples per packet is directly proportional to the sample rate and the elapsed speech storage time. Note however that D represents just the duration of the stored speech in each packet and does not include the time taken for speech to travel from its source to its destination.

The total time taken for speech transmission T must also include the transmission path delay P and can be expressed as follows;

$$D + P \leq T \leq 2\ D + P \qquad (4),$$

where P depends on the number of nodes in the ring, the delay at each node and the transmission medium used to communicate between nodes. The reason that the contribution towards the total delay made by speech storage can vary between D and 2D is that a queue exists at both ends of the transmission path. At the transmitting end of the link, the queue has a constant length, and is fixed by the packet size. At the receiving end, the queue length will vary to accommodate individual epoch synchronisation.

To estimate the best return of bit rate for a given speech delay, the overheads of each packet must be taken into consideration. These will have a decreasing effect as the number of samples in each packet increases. As an example, assume a constant overhead size of 32 bits and a ring capacity of 80 nodes. As the number of 8-bit samples per packet is increased, the overall contribution of the overhead to the data rate will diminish. This is shown be graphically in figure 6.2.

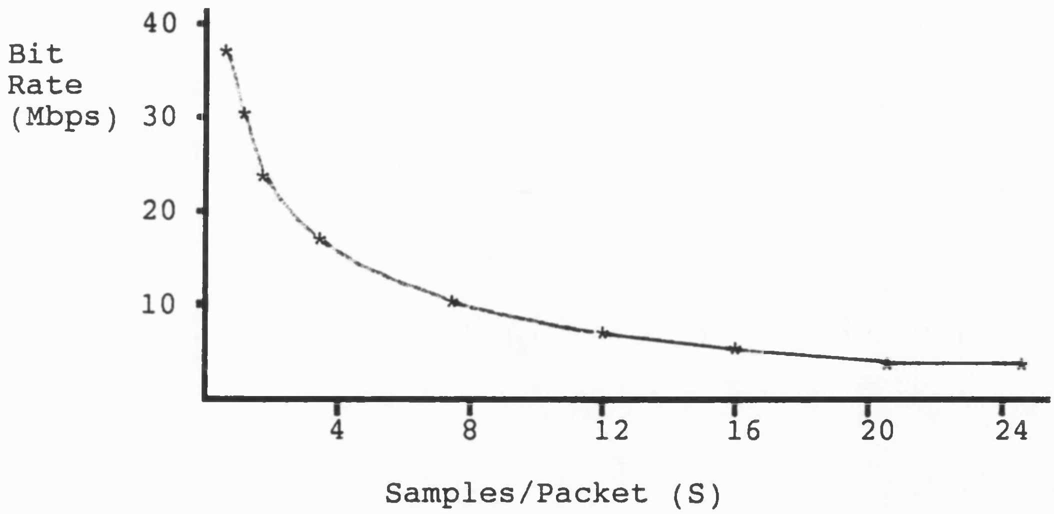


Figure 6.2. Bit Rate Optimisation.

If 1-bit rather than 8-bit samples were shown in figure 6.2, the discrete points would tend to merge, hence the curve is shown as a continuous line.

As the length of packets used in speech systems must be relatively small, the contribution of overheads towards packet size must be minimised.

6.2.2 Control Bus

It is possible to separate the transmission of control and audio data onto two buses or to integrate them onto one.

In several military aircraft that are already in service, and in some that will enter service soon, a control bus is already installed. It is the MIL-STD-1553 bus [34,35]. However, this is a centralized bus limited to a maximum size of 31 nodes which, because of its operation contains knowledge of system configuration in the central unit and therefore is less suitable for the distributed, expandable solution described in this thesis.

Integrating control and audio data onto the same network will save on wiring, installation etc. and will not present a large penalty in network bandwidth because the quantity of control packets will be much lower than the quantity of voice packets. Recent development of communication control systems (CCSs) utilising the MIL-STD-1553B control bus has shown that a 1 MHz bandwidth is ample for CCS control requirements. This should be compared with 16.8 MHz that would result from an 80 node system allocating 12 kHz per channel, transmitting samples in Manchester code and using the same percentage of overheads. Types of control packet will be discussed in section 5.2 of this chapter.

6.2.3 Model Decomposition

An initial thought for decomposing a design from the nodal data flow diagram in chapter 4 was that a large hardware implementation of each node would be necessary. Speed requirements initially dictated the use of hardware rather than software for voice data sorting and storage, and for the fibre optic network interface. However, upon further investigation, state-of-the-art processors and gate arrays were found to be able to perform well enough to render the consideration of hardware constraints unnecessary when designing network operation and nodal access of the network.

The data flow model forms the basis of the node design. Fast hardware registers can be provided for data to access the ring, and a packet structure can be accommodated on the network.

6.3 Functional Description

The Russell Ring was designed to support a system using very few types of units to form a communication control system (CCS). The ring operates by transferring information through every unit (node) connected to the network (see figure 6.3); a full cycle of the ring is completed when the originating node receives and removes its transmitted information (a packet). This full cycle of the ring rather than having the destination remove packets, is attributable to two reasons associated with intercom operation. Firstly, there are multiple destinations making it difficult to assign a particular node to remove the packet, and secondly, once the packet has returned to its source, it can be used to regenerate side-tone, thus giving the operator full confidence in the operation of the ring network.

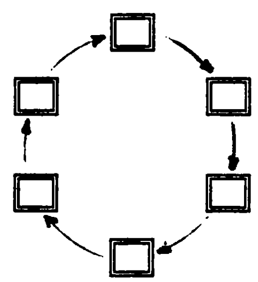


Figure 6.3. Direction of data flow.

6.3.1 Redundancy

To make any network more reliable, redundancy can be incorporated. This can be achieved at any layer in the design, the lowest being to duplicate the transmission medium and the means of access to it in the physical layer.

With a ring network, two variants of physical layer redundancy are possible, they contain uni-directional or contra-rotating data flow. The contra-rotating ring was chosen for reasons that will be discussed as part of chapter 9 (reliability and integrity).

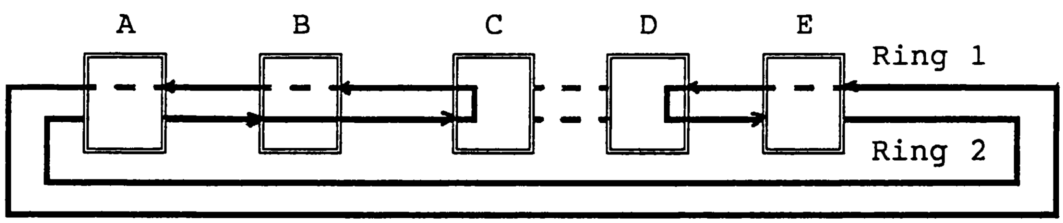


Figure 6.4. Contra-rotating ring with loop-back

Figure 6.4 shows the connection of a contra-rotating ring. The configuration is such that the effect of any single point failure can be removed by the automatic selection of switches for the purpose of re-routing the data around different parts of the network. Loopback is the process of transferring data from one ring to the other and passing it back to the previous node. This is shown in nodes C and D. Bypass is the process of passing data straight through a node bypassing the failed elements in that node so that the data will not be affected by any fault which may have occurred in the sub-system of that node. This is shown in node B.

The use of fibre optic point-to-point links dictates that signals will be converted between electrical and optical states at every node. The effect of failures in this process must be averted. The use of the loopback technique will overcome transmitter failures at a node. This is however, a matter of integrity, and therefore will be discussed in chapter 9.

6.3.2 Joining the ring

It is essential that each node remains passive on the ring network until it is appropriate for that node to join the operation of the ring, without causing any data loss to those nodes which are already operating on the ring. A separate power-up procedure is required before any node can participate in the rings steady state operation.

Joining the ring is a network specific feature and is only covered by the bottom two layers of the layered model approach. They correspond to;

Data-link layer	Check data, synchronisation, selection and de-selection of access registers.
Physical layer	Convert and decode network data, establish clock.

6.3.3 Ring Specification

The specification of the serial data transfer between units is based largely on research carried out by Harrison [12]. However, use will be made of components that have become available since the publication of his dissertation.

For the purpose of demonstrating the fibre optic, contra-rotating ring in the laboratory, data is routed via a multi-mode optical fibre and terminated by 9 millimetre SMA connectors. A bit rate of 20 Mbps is used with a transmission coding method for serial data transfers conforming to 'Manchester II biphase'. This is preferred to the 4B-5B scheme recommended by Harrison [12] to simplify the clock recovery circuits. There is little to be gained by using 12.5 Mbps instead of 20 Mbps terms of component size and cost, but much to be gained by using Manchester code.

6.3.4 Ring Storage

The real-time requirements of a voice system mean that nodes require fast access to the ring network at or shortly after the time requested. A series of slots would offer this capability more readily than waiting for a token.

Application of the Cambridge Ring [40] relies on storage that is inherent in the network, for its slots. This can take the form of wire or component delay. A problem with relating the Cambridge Ring to the requirements outlined in this thesis is that it is not easily expandable. As the number of nodes connected to the ring in some aircraft applications may be as few as say 20, significantly less storage will be available than for an application with 80 nodes. The size of the available slot must be defined for the minimum expected storage, unless a method of dynamic slot allocation is used.

Dynamic slot allocation can be achieved in two ways. Either the ring contains enough knowledge and intelligence to increase the number of slots as the number of nodes increases or, as each new node joins the ring, it inserts a fresh slot thus ensuring that the number of slots is compatible with the number of nodes.

This method of allocating slots dynamically as new nodes join the ring provides the added advantage that it possesses known latency for any instant of operation. This latency is bounded by design parameters which include an upper limit on the number of nodes.

With permanantly circulating slots of fixed length, known storage capacity is available. The nature of this storage is still optional. Figure 6.5 shows two methods for a node to communicate with the ring network, (a) illustrates the medium access method preferred for buses and the token passing ring, and (b) shows a method where all data passes through all nodes. The Russell Ring makes use of method (a) at power-up or when a node is required to be passive, and method (b) is used when the ring is operating actively in steady-state.

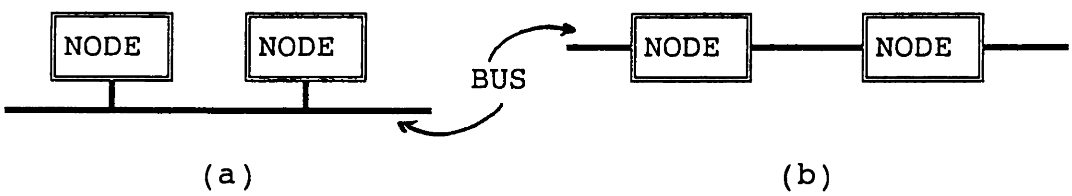


Figure 6.5. Methods of network interconnection.

The actual connection of registers from within the node to access the transmission medium is shown in figure 6.6. (a) corresponds to figure 6.5a in which data in the receive (Rx) register has to be monitored very quickly so that if the data is required by that node, it can be stored before the next packet arrives, and (b) corresponds to figure 6.5b where the register forms an active part of the ring.

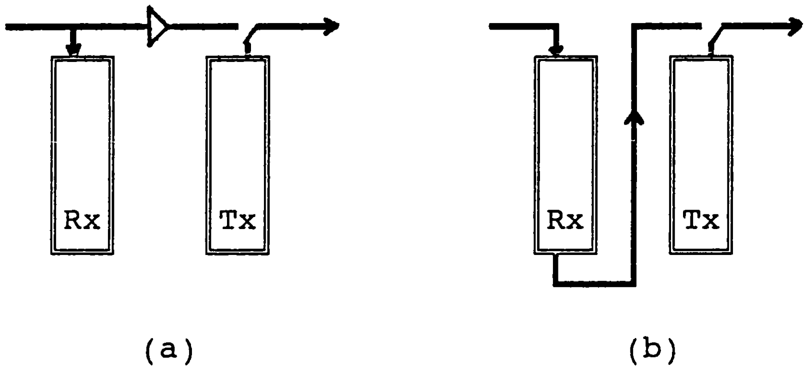


Figure 6.6. Medium access by registers.

6.4 Data Related Variables

Although factors affecting packet size and efficiency have already been discussed in section 2.1 of this chapter, a summary of some other relevant variables is given below:-

Distortion
Stored speech (D)
Bit-error-rate (BER)
Packet-error-rate (PER)
Samples per packet (S)
Number of nodes (N)

Bit rate (R)

Sample rate (F)

Sample size

No. of simultaneous receptions

Efficiency (data vs over-heads)

Packet size (L_p)

Gap size

6.4.1 Packets

From the graph in figure 6.2, section 2.1 of this chapter ("packet size" verses "bit rate"), it can be seen that having more than 12 samples in each packet returns little saving in bit rate. For reasons that will become apparent later in this thesis (specifically in chapter 7, audio quality), the length of speech required in each packet is approximately 2 ms. This gives a figure of 24 x 8 bit bytes of information per packet. It will be shown in the following section that 4 bytes of overhead is sufficient. This part will be known as the header of the packet. The information section of each packet contains either digitised voice or control data depending on the required function and is described later in this chapter. The header part of each packet may be considered as 4 separate fields.

6.4.1.1 Packet Header

The first of the four header bytes is reserved for the synchronisation of packets arriving at a node. As voice coding schemes may generate any code between all zeros and all ones, no unique code remains for the purpose of synchronisation. If a 4B-5B block code had been used, such a code would have been available [39], subject to clock recovery penalties. Use can be made however, of a gap which will occur between packets, so that each node may count the number of bits in a packet, wait for the gap, then look for an expected code, say all ones. This will provide enough transitions for the clock reconstitution circuits to lock.

The second of the four bytes in the header forms the control field. This field is specific to the system application. These bits have the following definitions:

Bit 0	-	Monitor Bit	-	Normal State 0
Bit 1	-	Full/Empty Bit	-	Full State 1
Bit 2	-	Data/Voice Bit	-	Data State 1
Bit 3	-	T/R PTT Bit	-	Keyed State 1
Bit 4	-	Intercom PTT Bit	-	Keyed State 1
Bit 5	-	Overide Bit	-	Keyed State 1
Bit 6	-	PCM/CVSD	-	PCM State 1
Bit 7	-	Parity	-	Odd

The functions of most of the bits in this field will become apparent later in this chapter, but briefly, the monitor bit has the function of allowing stray packets on the network to be eliminated. The full/empty bit indicates whether the packet is full or empty, informing each node when it is appropriate to insert a new packet. The data/voice bit allows nodes to differentiate between those packets containing voice data and those containing control data. Bits 3-5 in this field are specific to the operation of a CCS. Bit 6 is offered to the network user to run two different coding schemes on ring concurrently, and parity has its normal function, but applies to the header part of the packet only.

The third of the four header bytes contains a binary code indicating the unique address of the originating node, ie. the packet source address.

The fourth of the four header bytes contains the special address. This field may have different functions depending on the mode of operation. In the exceptional case when more than 256 unique source addresses are required, some bits in this field may be used as an over-flow. This gives an absolute maximum of 65536 nodes.

When the packet concerned contains voice information, a logic level one on bits 0-3 of the fourth header byte selects the intercom net of the same number. This provides for a facility where any or all of the four intercom channels can be used by each operator. The remainder of the field may be implementation dependent.

For control packets, this byte will perform the request, respond and confirm operation of the facility for automatic radio management, the coding for dynamic reconfiguration and the request and response for BITI. An explanation of these features is included later in this chapter but for completeness, the bit assignments are included below.

0000 0000	RAD_RQST
0000 0001	RAD_RESP
0000 0010	RAD_CONF
0000 0011	BITI_RQST
0000 0111	BITI_DATA
1111 1110	OK_PACKET
1111 1111	LOOP_BACK

6.4.2 Bit Rate

The bit rate of a network is inclusive of several features. Firstly, an analogy with the "30-channel PCM" [12] bit rate (R) shows that it consists of the number of slots per frame (N) times the number of bits per sample (usually 8) times the frame repetition rate (F). Therefore for "30-channel PCM":-

$$R = 32 \times 8 \times 8000 = 2.048 \text{ Mbps} \qquad (1)$$

For the Russell Ring, this formula would be inappropriate as it does not take into account the packet size including the header. In terms of the number of bits and the duration of transmitted speech, the bit rate is:-

$$R = \frac{N}{D} L_p = \frac{80 \times 224}{0.002} = 8.96 \text{ Mbps} \qquad (2)$$

The equation in (2) still has a further omission compared with the original analogy in (1), that is of the control and signalling overheads. The 30-channel digital telephony scheme has two slots for control and signalling thus bringing the total to 32 slots, this is an allowance of 6.25%. The same increase in (2) would give 9.52 Mbps. To provide a tolerance and to round R to an easily manageable number, 10 Mbps will be used. This provides for 1.04 Mbps of control data; twice that allocated if a seperate MIL-STD-1553 control bus were used. Note that no allowance is made at this stage for a line or block code. The use of Manchester II biphase will double the bit rate to 20 Mbps.

The maximum bus size for the Russell Ring is a variable that is governed by bit rate. Having shown that the proposed bit rate is suitable for an 80 node ring, the effects of using different bit rates and different numbers of nodes will be discussed in chapter 10 (performance). Similarly, distortion depends on the speech digitizing scheme that is incorporated and is therefore the subject of chapter 7 (audio quality).

6.5 Ring Operation

The signal on the ring consists of a sequence of slots each separated by an inter-slot gap. Each slot occupies a space of 28 x 8 bit bytes (224 bits) and when full contains one packet.

Each gap occupies the space of 32 bits. This gap provides adequate tolerance for any phase jitter that results from connecting a multiplicity of nodes in a ring. It also allows enough time for passive nodes wishing to join the ring actively to select their appropriate switches without causing the loss of any data already travelling around the ring. Finally and perhaps of least significance, using a 32 bit gap gives a total duration of packet plus gap of 256 bits. This is a convenient number for the implementation of binary counters.

Each node can use the data on the ring as it passes through the node if the data is required by that node. Data generated within a node is held within that node until an empty slot is transmitted from the previous node. A packet inserted onto the ring circulates until it returns to its source at which point the packet is replaced by an empty slot.

6.5.1 Monitoring Packets

In order to prevent a build up on the ring of packets which have in some way had their source address invalidated and hence been dis-owned by their originating node, it is necessary to check for damaged packets and then remove them. This is achieved by appointing a monitor node. All nodes are capable of being the monitor node. The monitor node is assigned as the first node on the ring to be powered up. However, should this node fail, a restart procedure is available to reassign the monitor mode function. The monitor bit in each packet will be changed to its active state by the monitor node. If any packet arriving at the monitor node is found to have its monitor bit set to the active state, thus indicating that it has failed to be removed by its originating node, then that packet will be replaced by an empty slot.

Each node also monitors the reception of its own returning packet for the monitor bit being set. If it continually receives unmonitored packets, that node initiates a procedure to reassign the monitor function. The initiating node sends out its next packet with the monitor bit set. If the packet does not return, then the ring is functioning correctly and that node has merely lost 2 ms of speech (a different node may have completed reassignment first). However, should the packet return, the initiating node must adopt the monitor function. If during the course of reassignment, the initiating node receives a packet with its monitor bit set and a lower address than its own, then the initiating node must yield to avoid having two monitor nodes on the network.

One advantage of this monitor reassignment routine is that very little network usage time is lost because each node is using packets that would be transmitted anyway.

Monitor mode operation only corresponds to layers one and two (physical and data-link) of the 7-layer reference model, and is therefore invisible to the rest of the system except for error reporting.

6.5.2 Voice Packets

A voice packet is indicated by the appropriate bit being set in the control field of the header. The information section of a voice packet contains 192 bits of digitally coded audio amounting to an elapsed time of approximately 2 milliseconds. The time limit imposed on each packet is derived from delayed auditory feedback figures quoted by Carpenter [14] and the time taken to construct a packet from stored samples, together with subjective tests performed in the laboratory using a Mullard TDA 1022 analogue delay line.

There exists the possibility that voice packet overload may occur. This may be caused when the ring is operating beyond its design capacity. Packets arriving late at their destination are useless because they must be decoded in sequence and within real-time constraints, therefore, if the packets have not been transmitted onto the access medium from their source within a pre-determined period, they must be scrapped.

6.5.3 Control Packets

A data packet is indicated by the appropriate bit being reset in the control field of the header. The information field in these packets will, in normal use, provide the facility for selection, control and tuning of radios and encryptors. The other uses for data packets will be for down-loading built in test information (BITI) and for the reconfiguration of a damaged ring.

6.5.4 Radios

The selection of radios is achieved by a method of request and response. The requesting node sends a data packet which defines the particular facility that the operator requires (in terms of frequency, power, antenna, etc.). Each node able to respond to such a request (ie. radios) will act by either setting the failure-to-comply flag in the packet and re-transmitting it to the next node or by accepting the request and responding with the parameters that need to be set up.

When the originating node receives the packet with the failure to comply bit is set, the node will transmit a warning to it's control panel indicating that no connection was possible or if a connection between the two nodes has been made it will transmit a fresh packet to the responding node to confirm this.

The responding node from that point on will read all packets originating from the requesting node, however, they will only be converted from digital to audio if the T/R PTT bit of the control field header is set to the keyed state and if the set-up procedure outlined in the previous paragraphs has signified to the radio that the operator requires receive and transmit rather than just receive. If the operator changes his requirements (eg from receive only to receive and transmit), a new request is made to the radios. The radio previously selected is most likely to be the one to respond, but this is not essential.

6.5.5 BITI

Built In Test Information (BITI) is transmitted onto the ring network from any node when it is requested via a data packet from the network. This information is the result of continuous monitoring by each node, of its on-board circuit status and the state and history of the reconfiguration mechanism on the ring, as well as any BIT outputs from radios, encryptors etc.

The format of the BITI request and the BITI response data packets is controlled from the special address field in the header of each packet.

BIT is a continuous function which stores an updated data base in each node. The contents of this store may be forwarded to a requesting terminal in the form of BITI. The requesting terminal may take the form of either an operator control panel or a communications services monitor (CSM).

6.5.6 Dynamic Reconfiguration

Under normal operation, only one of the contra-rotating rings will be used. In the event of a fault, illustrated in figure 6.7 between station 1 and station 2, the node at station 2 fails to detect a signal from the previous node (station 1) or if a constant stream of incorrect data is detected from station 1, then station 2 will select its loopback switch and transmit a reconfiguration packet to station 3, then wait a short time for a response from station 3. This procedure will continue around the ring until a failure to respond is encountered (station 1 for the example in figure 6.7), at this point a second loopback switch is engaged and the ring network is once again complete.

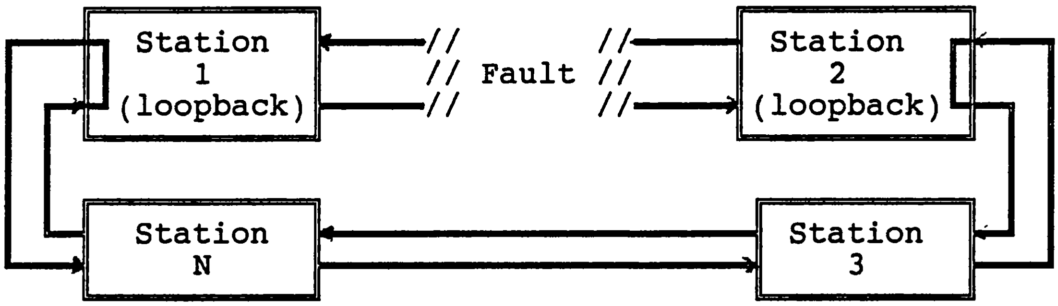


Figure 6.7. Dynamic Fault Exclusion.

The packet that station 2 transmits after setting its loopback switches (loopback switching is covered in section 6.4 of this chapter) is distinguished by having all ones in the special address field. This type of packet (the loopback packet) is passed straight onto the next node on the inner ring by each node that receives it. Nodes only have one receive and one transmit register, therefore the transmit register is switched onto the outer ring and the node responds to the one it received from, with an "OK" packet (again, distinguished by its special address). If a node fails to receive an "OK" packet within a predetermined time, its loopback switches will be operated.

This section has introduced the concept of reconfiguration reactions in the event of a fault or failure.If as a result of the reconfiguration process it is found that a fault has occurred in a node rather than a failure of a fibre optic link, then the bypass switches in that node may be selected to eliminate that node from the operation of the ring network and the loopback switches may be reset. This is however, subject to the degree of failure within the node (see appendix IV).

The reconfiguration process covers all four of the transport layers in the 7-layer reference model [13]. A decision on which ring(s) the registers are connected to, and the formatting of new packets for transmission (loopback packet) cannot occur below the transport layer (layer 4). This constraint is imposed to maintain a constant method for information handling and the flow of data, this process should comply with all other usage of packets.

6.5.7 Power Up

The system protocol at start up is as follows. When a node is first switched on, it continues to act merely as an electrical repeater station without introducing any delay. When an inter-slot gap is detected, a register is inserted into the data path in such a way that no data is lost or corrupted. The new node then behaves as previously described. If a node being powered-up is the first node to join the ring, it becomes the monitor node.

It is assumed that when the aircraft power-supplies are switched-on, a low current is supplied to each node to support the electrical/optical conversion components. This subject is covered in more detail under integrity and reliability in chapter 9.

6.6 Practical Design Considerations

Those problems associated with designing a protocol that will operate in a "real" system will now be considered.

6.6.1 Clocking

In connecting several registers in series and in a ring, the continued generation, reconstitution and regeneration of the clock (see figure 6.8) may cause phase jitter and frequency slippage. This may of course lead to the loss of data due to decision point movement if that phase jitter becomes to large. Several possible solutions to this dilemma now arise. Firstly, clocks in all nodes could be made accurate and not locked. To reduce the probability of errors to an acceptable level however, would require a clock at least three times the size of the rest of the node [47]. Secondly, if the receive clock was locked onto incoming data and the transmit clock was free running errors would still occur, unless thirdly, the start of a transmit packet sequence was inhibited until a point marked by the nth bit of the receive packet sequence (see figure 6.9). This also has the effect of creating a gap between packets which will absorb fluctuations in phase and frequency. These fluctuations, if not countered would have the effect of "bunching" and "spreading", which both lead to a loss of data.

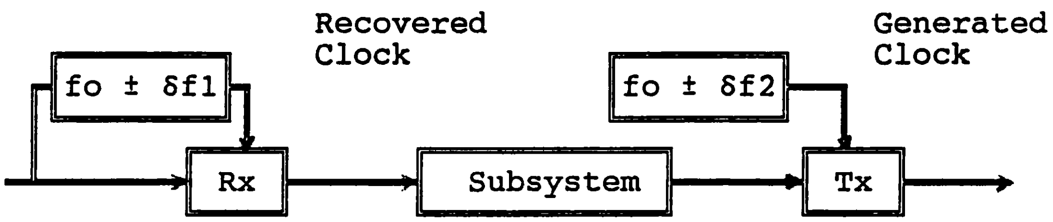


Figure 6.8. Notional Clock System.

In the third scheme, data can still be reconstituted correctly as it is decoded using the recovered clock. Epoch periodicity is maintained as this may also be driven by the recovered clock. Frequency and phase deviations have little effect on the network because each independent, point-to-point-link clock is synchronised in phase and frequency, once every epoch (frame), rather than every bit.

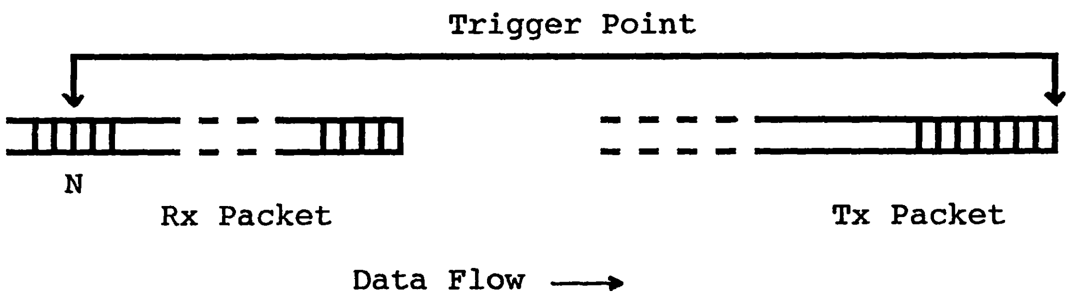


Figure 6.9. Timing Solution.

The network must be capable of having new nodes added to it during normal operation with no adverse effects. This is especially important in an aircraft environment. It is achieved by each new node joining the ring's operation during a gap after first having locked its clock recovery circuit to the incoming data.

6.6.2 Joining at Power-Up

All nodes at power-up will transmit a blank packet with their own unique source address. If the next packet to be received from the network is the same packet, then that node is the first to join the network and hence should adopt the responsibility of monitor node. The monitor node will take responsibility for removing all damaged or lost packets.

If the new node is not the first to join the ring, a gap search procedure is initiated. The actual connection of the node to the media must be made with no significant disturbance to an already operating network. This is achieved by monitoring the medium, detecting the synchronisation field of a header, counting the length of the packet for the occurrance of a gap between two packets, and then all connections to switch the node registers into the path of the network can be made without causing the loss of any data.

6.6.3 Integrity

There are three levels of self test within nodes. The first is side-tone. This is the operator's facility to listen to what he is saying. It proves that the ring network is operational as the digitised speech packets would have made a full cycle of the ring before being received back at the originating node.

The second level of self test is continuously operational and consists of monitoring hardware and software internal to the node and also continuous monitoring of network activity. The built in test information (BITI) that is a result of continuous monitoring for faults and failures is used by the node to determine whether it is necessary for that node to leave normal ring operation (this can be achieved by the selection of by-pass switches) and is used in the third level where it is available for transmission on request to a control panel or via the ring network to any other node (such as a system monitor - not to be confused with monitor mode) that may request the information. This facility will greatly improve service and maintainance times and for military aircraft will improve the turnaround time between missions.

6.6.4 Self Healing

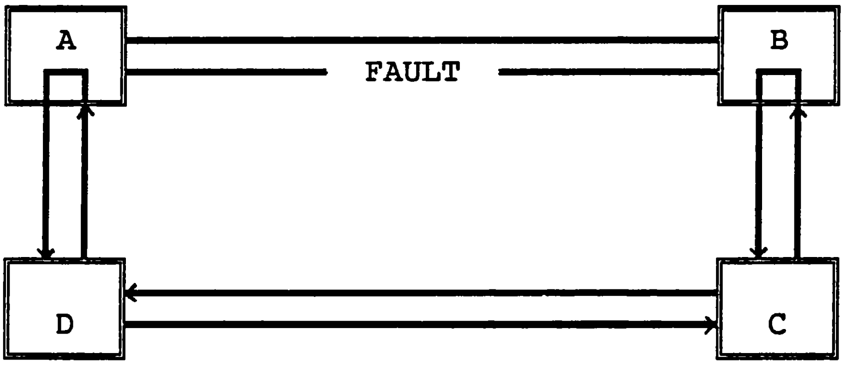


Figure 6.10. Self Healing Network.

This section gives a more detailed example of the network reconfiguration process. It will however, be discussed again in chapter 9 in connection with integrity and reliability.

Under normal operation, only one of the contra-rotating rings shown in figure 6.10 is used. However, if a node should fail, or a portion of the ring between nodes should fail, then means are provided for the dynamic reconfiguration of the ring.

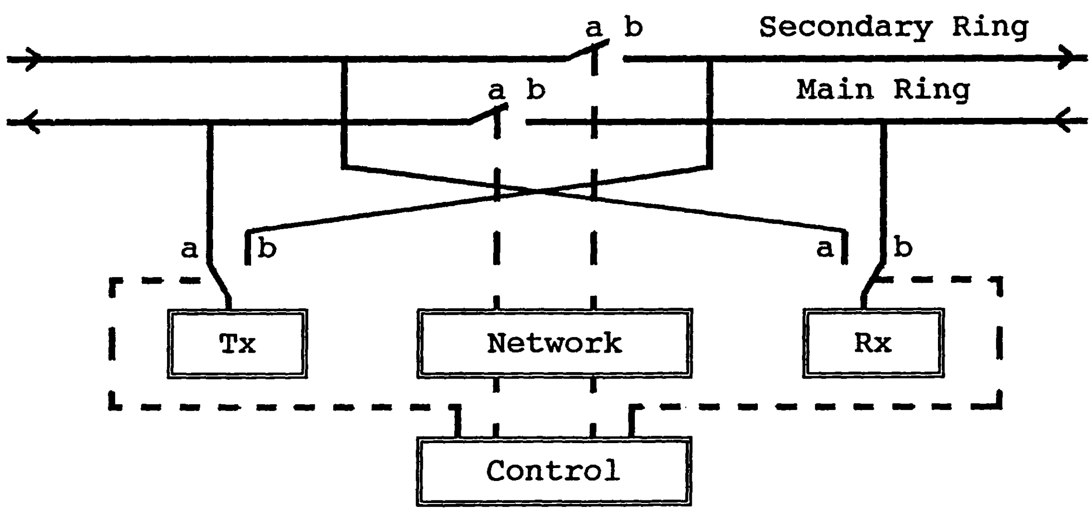


Figure 6.11. Reconfiguration Control

If a node receives a loopback-packet, it is effectively being asked to consider reconfiguring itself. To determine whether it should reconfigure itself to loopback from the main ring to its secondary output, it first needs to establish whether a fault is within its vicinity. In the event that loopback is required, the switches shown in figure 6.11 are changed so as to connect the transmit register (Tx) to the other ring. This loopback function can be performed either with an optical switch or electrically with the appropriate conversions between electrical wires and optical fibres depending on the technology that is available for a given environment.

With the ring operating in steady state, a node's medium access switches are in the positions shown in figure 6.11. On receiving a loopback-packet the node retransmits the same packet on the main ring. The node then selects the Rx switch to position "a" to wait for a response, and selects the Tx switch in position "b" to transmit the "OK" packet onto the secondary ring.

If an "OK" response arrives at the Rx register within the specified time, the Rx and Tx switches are reset to their steady-state positions and the secondary ring switch is set to "b". If there is no response to the loopback-packet, then the Rx switch is set to "b" and loopback has been selected.

Referring to figure 6.10 and assuming that the inside ring is the one being used in steady state operation, once node B detects the absence of a signal at its input, it transmits a control packet to node C. Node C then transmits the control packet to node D on the inside ring and a reply back to node B on the outside ring . This process continues until the sending node fails to receive a reply. In the example it is node A, which then selects loopback thus eliminating the fault between nodes A and B. A flow-chart of the nodes self-healing process is shown in figure 6.12.

Subsequent chapters of this thesis show how the Russell Ring can be implemented and how it performs.

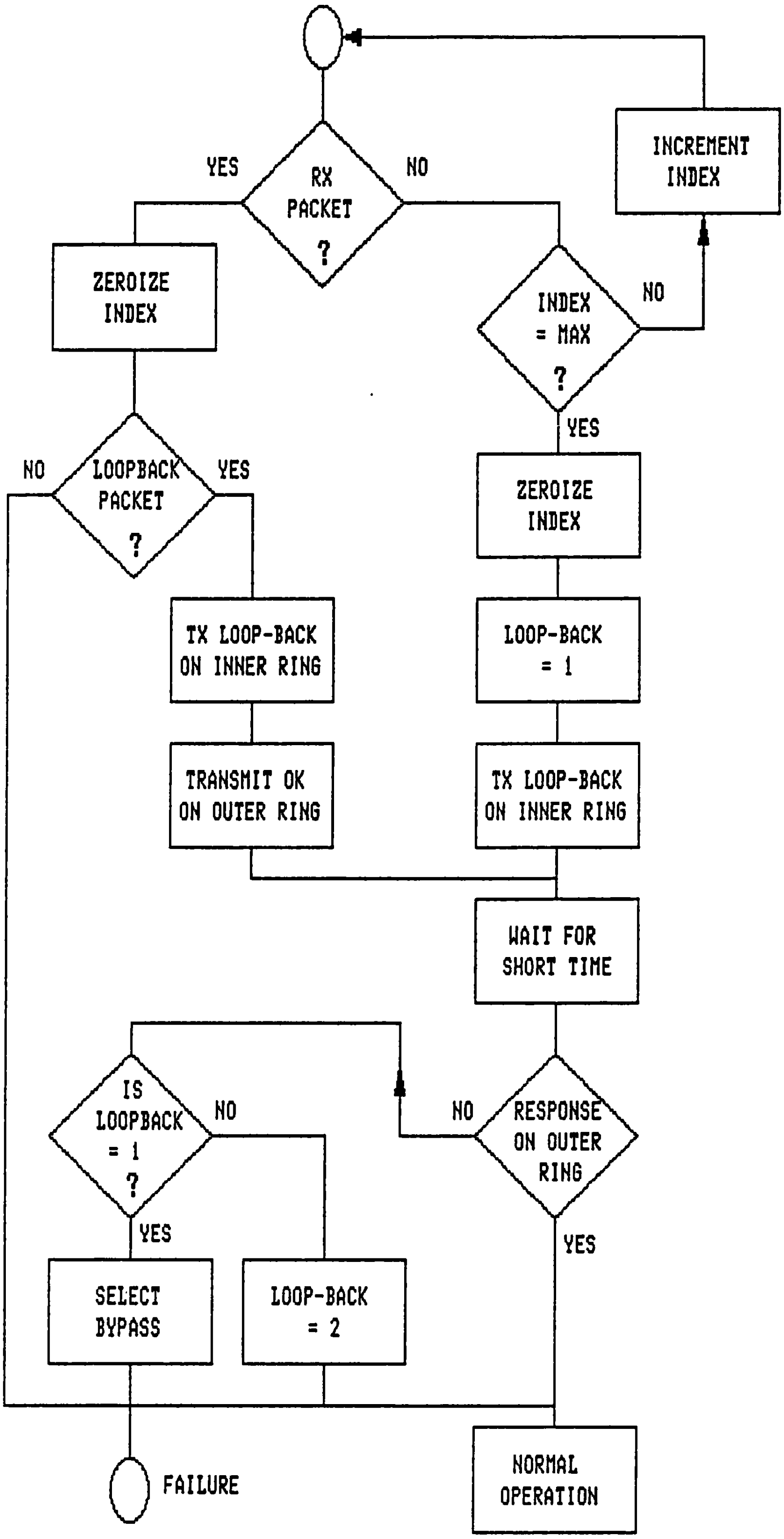


Figure 6.12. Flow-chart of self-healing operation.

7. AUDIO QUALITY

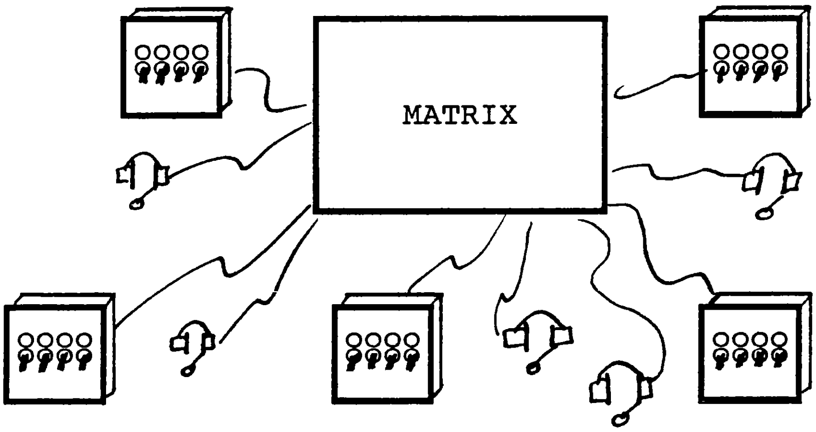
This chapter investigates speech quality through digital transmission systems in order to determine which method or methods of coding are most suitable for the particular application of an aircraft communications control system. A reduction in quality compared with conventional analogue systems is unlikely to be acceptable to potential users.

Assumptions are that the audio is required to be digitised for the purpose of time division multiplexing (TDM) and the audio quality will be comparable or better than existing aircraft analogue communications systems, eg. Robertson [19].

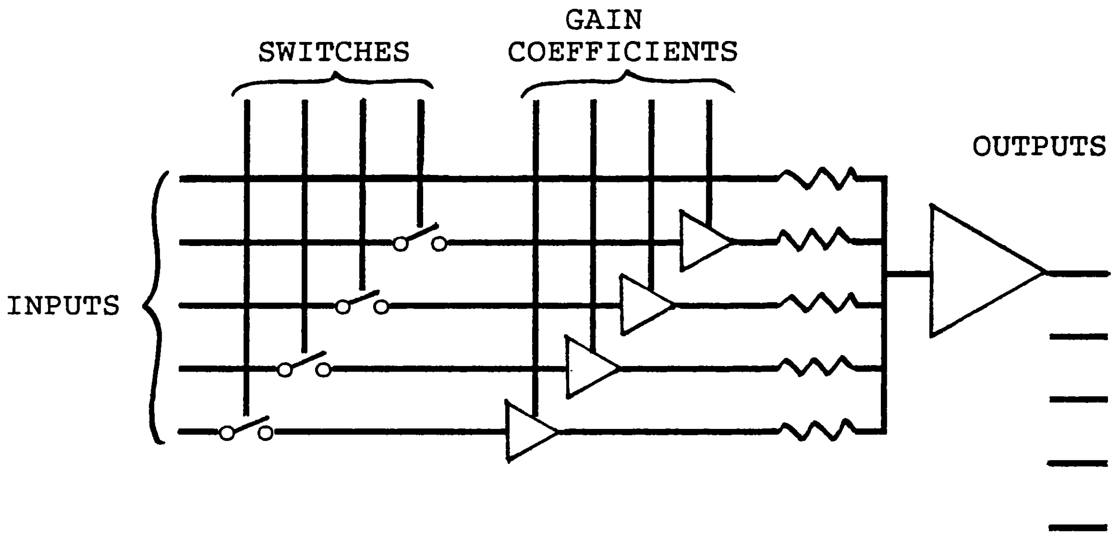
Considerable research work has already been done in the field of digital telecommunications [6,20] and several purpose built integrated circuits such as CODECS and MODEMS are readily available to satisfy the requirements of public and private digital telephone networks. However, the use of these in aircraft is limited by their availability in extended temperature ranges and their incompatibility with CCS audio bandwidths conventionally employed. The International Telegraph and Telephone Consultative Committee (CCITT) is the body formed to standardise on international telephone systems. CCITT have standards on speech and transmission quality, digital coding schemes and quality analysis tests among others eg. recommendation G712 [4].

Work by the Royal Aircraft Establishment [30], has shown that an operational bandwidth of 400 Hz to 5 kHz is needed for optimum communication. This is based on improved speaker identification in noisy environments as well as providing a quality acceptable to comfortably listen to for periods up to 12 hours. As a result, many VHF and UHF radios in current use on aircraft can transmit and receive an audio bandwidth of 5 kHz. It is clear that any reduction in bandwidth caused by the CCS will degrade the performance of the communication channel. The cascade nature of back-to-back systems is also a reason for requiring the wider bandwidth (ie. a scenario involving communication between two or more platforms with the same type of communication system). This will involve multiple digitization and de-digitization and hence this is equivalant to multiple cascaded filters which in an analogue system reduce the bandwidth. To illustrate this, two cascaded single pole 3.4 kHz filters (CCITT standard telecom bandwidth), will reduce the overall 3dB bandwidth to 2.14 kHz.

One problem which arises from designing a digital communications control system (CCS) to give similar facilities to that of the existing analogue systems is that each operator may require to listen to several audio channels simultaneously. In existing systems this was obtained by having large centralized audio boxes each with a switching matrix which could input from and output to all channels (see figure 7.1a). Combinations of channels could then be selected by switching the relevant signals through gain control amplifiers and into summing amplifiers as shown in figure 7.1b. In a distributed digital system there would no centralized units so that gain control and signal summing must be achieved at each audio interconnection, or node. There are two solutions to this problem. Either banks of digital to analogue converters (DAC) can regenerate the audio signals which can then be gain adjusted and the signals summed (figure 7.2) or, the functions of gain adjustment and signal summing can be performed digitally followed by a single DAC (figure 7.3), reducing the quantity of components considerably as the processing for multiple simultaneous reception of digitised audio signals can be performed in one digital signal processor (DSP).



(a)



(b)

Figure 7.1. Conventional methods of audio distribution.

(a) Centralized system. (b) Switching matrix.

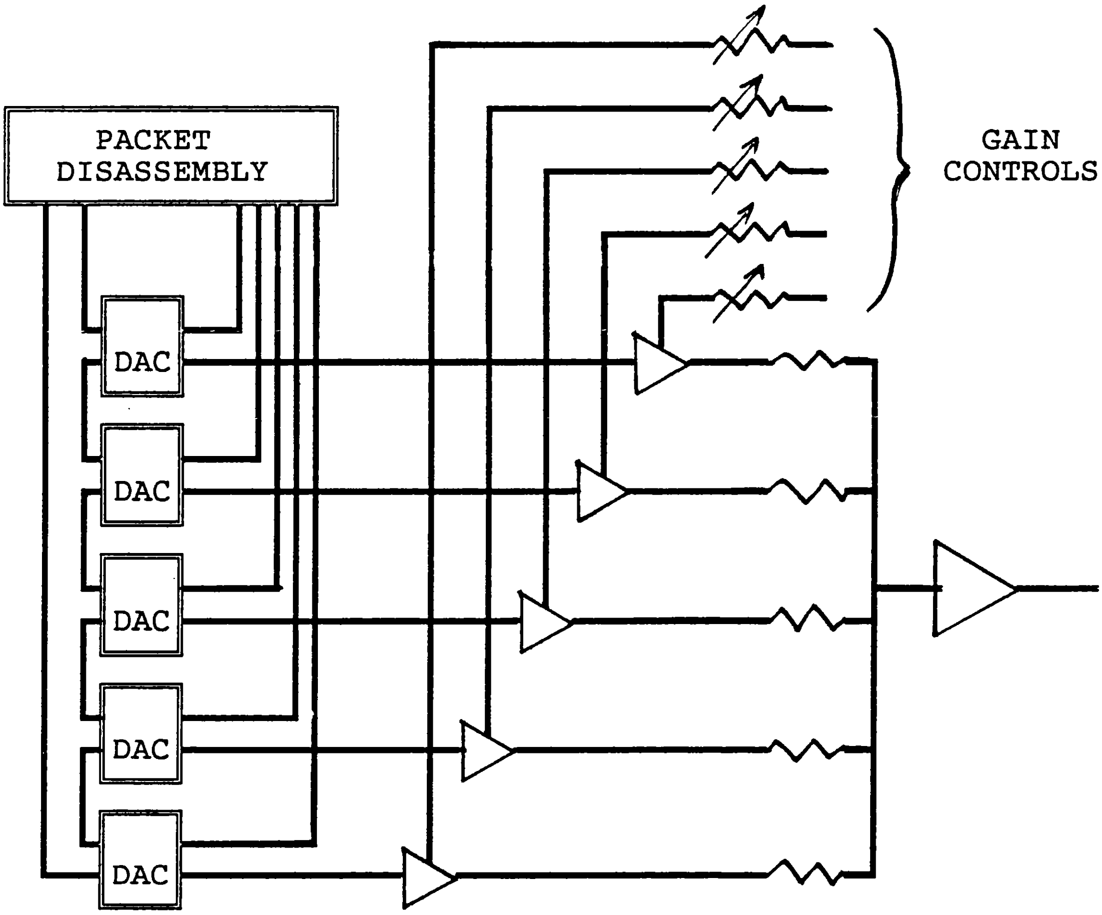


Figure 7.2. Multiple DAC decoding.

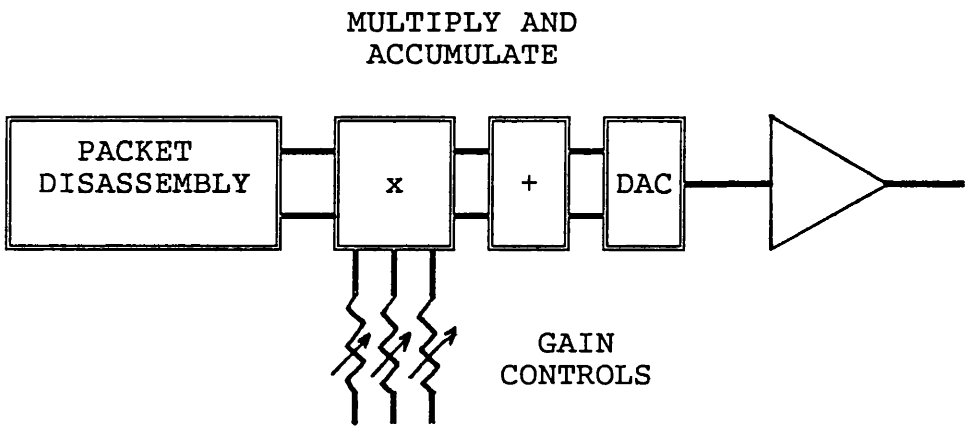


Figure 7.3. Single DAC decoding.

7.1 Audio Digitization Techniques

Most methods of processing or coding audio signals can be classified as belonging to one of two general systems, waveform digitization methods or vocoder methods.

In waveform coding, the speech waveform is sampled, represented by a digital code and transmitted to a receiver. At the other end of the transmission path, the analogue signal is reconstructed from the code to reproduce the original signal as accurately as the system will allow. With vocoder methods, speech is sampled and stored against a model of voice characteristics. These models may for instance represent the vocal tract and vocal chords etc. (figure 7.4 shows how a vocoder may reconstruct voice signals). Information on pitch and filter characteristics is transmitted so that at the receiver, speech can be reconstructed from a representation of its main elements.

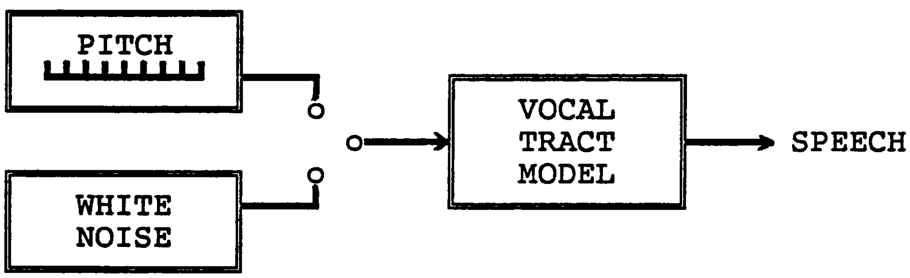


Figure 7.4. Vocoder recovery model.

7.1.1 Pulse Code Modulation

The first of the waveform methods to be considered is pulse code modulation (PCM). It is commonly accepted that the highest quality reproduction from digital coded analogue signals can be obtained by PCM. This is the most basic digitization system. In this system, the transmitted code is a series of binary numbers each representing a sampled amplitude. The digitization of the sample introduces noise, known as quantization noise, which is dependent on the quantization step size. Therefore the more bits that are used in each sample, the smaller will be the noise level. This quantization action also sets the dynamic range for the sampled signal, as the minimum amplitude of signal to be transmitted is equivalant to the step size. The dynamic range as well as the signal-to-quantization-noise ratio can be obtained from the number of bits in the sample (N).

$$\text{Dynamic range (dB)} = 20 \log_{10} (2^{N-1})$$

To obtain a range of 60 dB we need a sample size (N) of 11 bits assuming that the sampling device linearity and accuracy have no effect. Each additional bit increases the range by 6 dB and readily available analogue to digital conversion components operate at 8, 10, 12 and 16 bits. The serial data transmission rate for PCM is obtained by multiplying the sample size (N) with the sampling frequency (f_s).

7.1.2 Companded PCM

The logarithmic response of the ear with amplitude means that accuracy at high-amplitude signal levels is unnecessary. Applying a non-linear (logarithmic) law to the quantization step size enables the serial bit rate to be reduced without seriously impairing audio quality from that which would be obtained from linear PCM. Speech signals exhibit a wide dynamic range and differ considerably in mean power level prior to coding. The difference in mean power level, sometimes as much as 40 dB, can be caused by variations from person to person as well as differences in equipment, see Coates [21], pages 208-214. The non-linear compression and expansion of transmitted speech is known as companding. The CCITT have standardized on two possible formulae for companding. They are μ-law which is used in North-America and Japan, and A-law which is used in Europe. Here only A-law companding [4] will be considered. "A" represents the coefficient of compression which is shown in figure 7.5 and the following formulae represent the conversion process:

$$Y = \frac{1 + \ln(Ax)}{1 + \ln(A)} \quad \text{for } 1/A \le x \le 1$$

$$Y = \frac{Ax}{1 + \ln(A)} \quad \text{for } 0 \le x \le 1/A$$

where x = input / max input and
Y = output / max output.

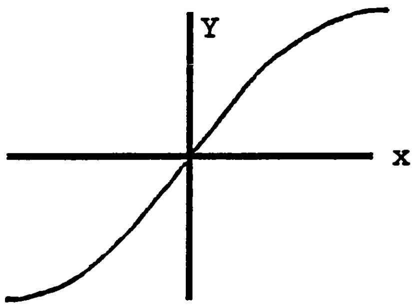


Figure 7.5. A-law companding, compression characteristic.

Companding can be implemented in either the analogue or digital sections of the subsystem. Using the above law, 12-bit samples can be reduced to 8-bit values (using A = 87.6) with no noticeable loss of audio quality. This is the method commonly employed by CODEC chip manufacturers (see appendix VI), providing a serial bit rate of 64 instead of 96 kbps (8kHz sampling). This is the rate which is to be used in the Integrated Services Digital Network (ISDN) and System X.

6.1.3 Differential Pulse Code Modulation

Reductions in bit rate generally involve a reduction in audio quality. However, audio quality does not reduce on a pro-rata basis with serial bit-rate, therefore reductions beyond companded PCM still warrant discussion. Differential pulse code modulation (DPCM) works on the assumption that there is a high degree of correlation between successive PCM samples, which makes future values easy to predict. This means that as each sample is quantised into a N-bit word for transmission, much of the transmitted data on each new sample is redundant. The serial bit transmission rate can therefore be reduced if the difference between the sample and an estimate of the sample built up from several preceding samples is encoded, rather than the complete sample. A general differential quantisation scheme is shown in figure 7.6. See Schwartz [22] page 142.

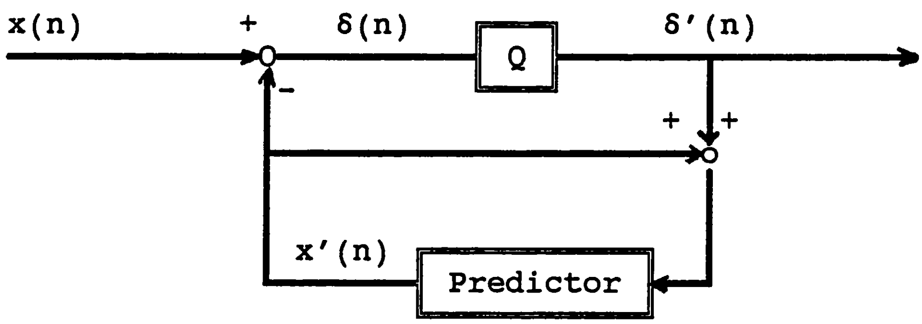


Figure 7.6. Block diagram of DPCM coder

7.1.4 Adaptive Differential Pulse Code Modulation

The basic system shown in figure 7.6 gives a reduction in bit rate over standard PCM but to obtain optimum performance the weights of the prediction [31] should be changed with time, forming an adaptive predictive scheme. Thus, the predictor must vary with time to cope with the changing spectral envelope of the speech signal as well as the changing periodicities in voiced speech.

Adaptive Differential Pulse Code Modulation (ADPCM) was developed as the next obvious extension from the two previous coding schemes. If the quantiztion steps of A-law coding were made variable then the system would be called adaptive. Combine this with DPCM and the result is a scheme as shown in figure 7.7 that can maintain the quality of PCM systems and the reduced bit rate of DPCM. The penalty for this however, is complexity.

The key element in the block diagram in figure 7.7 is the step size strategy. This element is not linked to the prediction, which can operate in the same way as DPCM, but is controlled by the variability of the signal. A better understanding of this will follow the discussion of step size associated with continuously variable slope delta-modulation (CVSD).

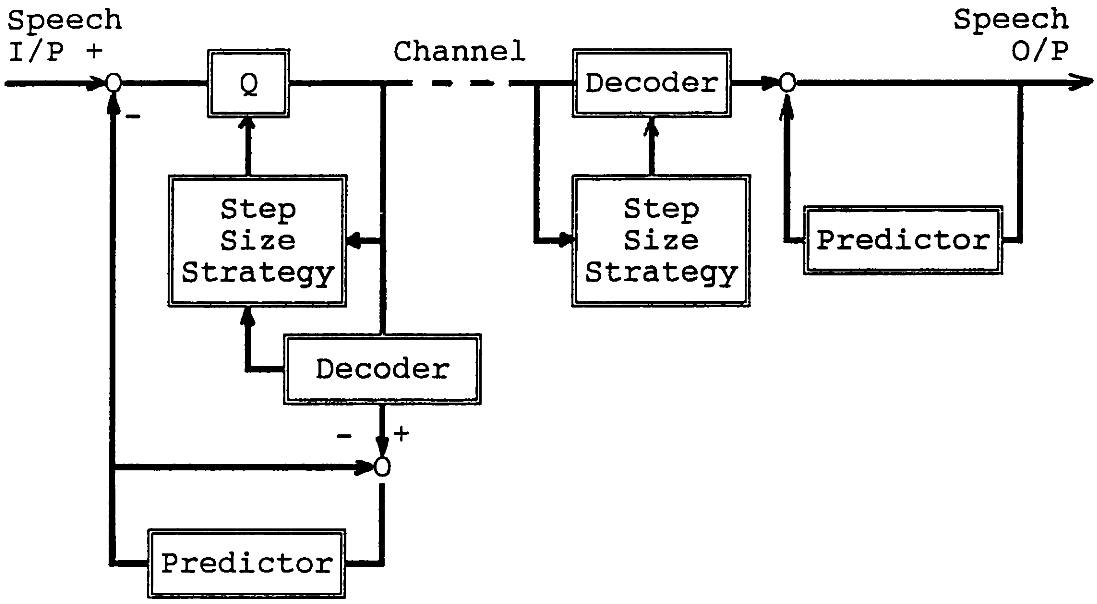


Figure 7.7. Block diagram of ADPCM coding scheme

7.1.5 Delta Modulation

Delta modulation (DM) and its derivatives are optimised for voice signals. DM is the result of DPCM taken to its extreme, only one digit being coded for the differential element. The sampling rate and output bit rate are identical in DM because it is a 1-digit code and therefore, to achieve performance comparable with a conventional N-digit PCM system, the sampling rate for the DM system must be increased substantially over that required for PCM. It is the 1-bit architecture which has caused some manufacturers to favour DM as the basis for "single-chip" coder decoder (CODEC) integrated circuits.

In the block diagram in figure 7.8, the 1-digit code can take on the value of + or - depending on the direction of the slope of the input signal with respect to the modulators record of the signal amplitude.

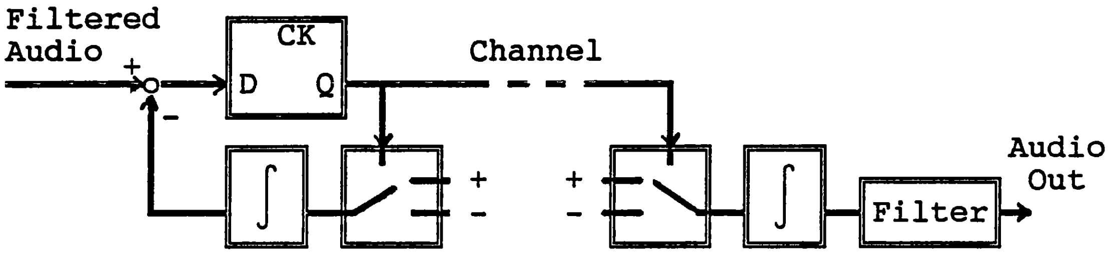


Figure 7.8. Block diagram of Delta-modulation.

Common problems associated with the use of Delta-modulation (DM) are granular noise and slope-overload. Granular noise occurs when the DM coder is tracking an input with rapidly alternating step polarities [44] and slope overload occurs when the input signal is changing amplitude to quickly for the modulator. The distinction between the two is very significant (both statistically and perceptually) and noticeable because of the coarse quantisation involved.

7.1.6 Continuously Variable Slope Delta-Modulation

The most common problem encountered when using DM is slope overload. The performance of the delta modulator can be substantially enhanced by dynamically varying the loop gain in sympathy with the average power level of the input signal. This system is known as continuously variable slope delta-modulation (CVSD). The loopback gain (quantization) is generally increased when three or four consequtive bits [23], depending on serial bit-rate, are all of the same value, and decreased in the presence of alternate 1s and 0s.

7.1.7 VOCODERS

There are several features of vocoder methods which make them less desirable for speech coding in an aircraft communications system. Firstly, the algorithms used for implementing vocoders are more complex than those used in waveform coding; because of the need for replication in a CCS, size and cost become major considerations. The quality of vocoder generated speech is also inferior to that of waveform coded speech [48,49]; this point is important for speaker identification as well as articulation. Vocoders are normally chosen as a method of conversion to meet severe channel bandwidth constraints. Bit rate and channel bandwidth however, are not major limitations in an intra-aircraft CCS, as it will not incorporate any existing communication channels and the use of optical fibres and associated technology make higher bit rates feasible.

For the sake of completeness, and to pose an alternative to waveform coding if the limited bit-rate requirement were to become necessary to meet applications other than for aircraft, a brief explanation of two vocoders follows.

7.1.7.1 Adaptive Transform Coding (ATC)

ATC involves block transformation of windowed input segments of the speech waveform. Each segment is represented by a set of transform coefficients, which are separately quantized and transmitted. At the receiver, the quantized coefficients are inverse transformed to produce a replica of the original input segment. Successive segments, when joined, represent the input speech signal.

7.1.7.2 Linear Predictive Coding (LPC)

Another method of representing the spectrum of a speech sound is by means of linear predictive coefficients. In essence, an LPC vocoder is an ADPCM system (see figure 7.7) in which the prediction residual is replaced by the pulse and noise sources as in figure 7.4.

7.1.8 Suitability of Coding Schemes

To provide a comparison between the different methods of waveform coding, signal-to-noise ratio (SNR) can be used. The result of this comparison depends on the bit rate being considered. A system that functions best at one bit rate is not necessarily the best at another. An illustration of this is shown by Flanagan [44] in figure 7.9.

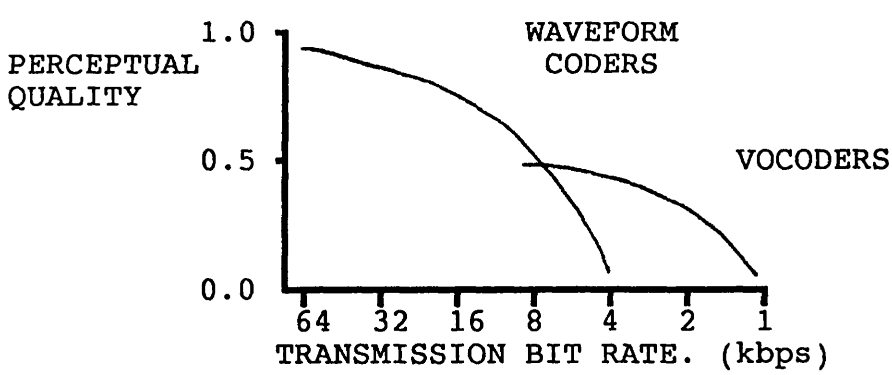


Figure 7.9. Variable rate coding configuration.

The selection of a suitable coding scheme must inevitably be a compromise between quality, complexity and coding rate. Therefore, the choice of coding scheme will reflect the application and its requirements. The system being considered in this thesis requires relatively high quality for ergonomic reasons [30], and a level of complexity which depends on the size of its implementation. Coding rate however, is of less significance than the other two constraining factors because of the use of good quality transmission paths.

7.2 Comparison of Techniques by SNR Analysis

The intelligibility of speech is essentially a subjective phenomenon. To this end, the measurement of how well a speech system has performed has in the past been made using subjective tests with talkers and listeners using sentences, rhyme words, or other test material. This approach has the advantage of obtaining a direct comparison. However, there are some serious drawbacks, such as the time taken to perform these tests and the poor information available from qualitative results, on why some communication channels perform better than others.

7.2.1 Analytical Results

The digital techniques described in this chapter need to be analysed to investigate their relative performance for given sets of circumstances. Although some of the techniques can be shown to perform well in their favoured environments, an analysis is required to show how they perform over the whole range of environments that might be encountered with aircraft systems.

The preferred 'bench mark' for comparing the performance of communication channels is signal-to-noise ratio (SNR) [23]. The noise caused by digitizing a waveform over a period of time will now be investigated.

The noise added by each technique differs in nature. PCM adds noise caused by the difference between the actual sample value and the level which is assigned by the encoding system. This is illustrated in figures 7.10. The following formula can be used to obtain the maximum signal-to-noise ratio (see Schwartz [22] page 119):

$$SNR = 4.8 + 20 \log(2^{N-1})$$

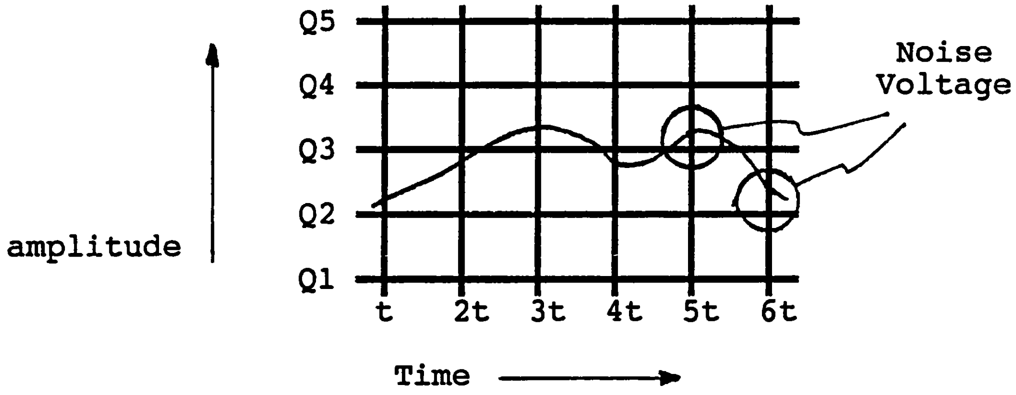


Figure 7.10. Quantising noise and quantising levels

Using a companding technique, the analysis of SNR should be based on the nature of speech signals. The actual signal appearing at the input to the sampler is likely to be significantly less than the peak value. From Schwartz [22] page 152, the input signal power level follows a binomial distribution. If the mean input power is 1/4 of the peak value (6dB deviation of speaker's power level), the probability of the input signal falling within the sampler range is 99.994% [24]. (Theoretically the gaussian variable can take on any value whatsoever). If we then pick a peak amplitude (A) that is four times the mean signal strength (μ) so that $A=4\mu$, the compander characteristic can be designed to give a relatively flat response for SNR across the amplitude range in most use. The system will perform better than linear PCM on the basis that SNR for the linear version gets appreciably worse for small signals. See the graph in figure 7.18.

Noise generated by delta-modulation (DM) adds an extra element to that generated by PCM techniques. This new element occurs when the input waveform changes amplitude quicker than the sampler can track it.

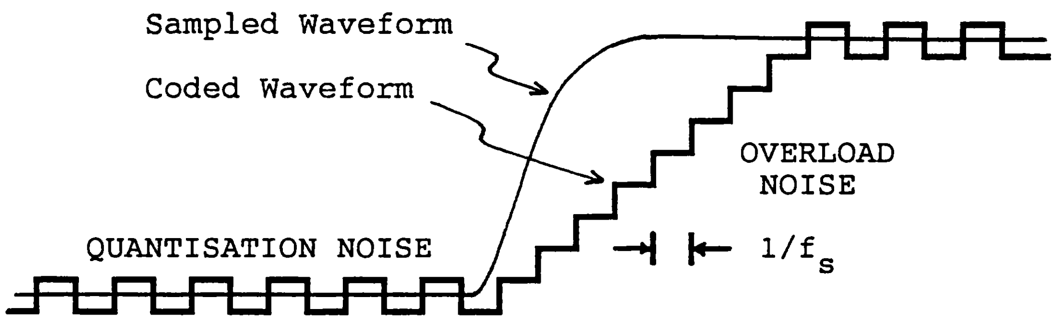


Figure 7.11. Slope overload

The diagram in figure 7.11 shows an area of slope overload. This leads to a portion of noise which can be much larger than quantisation noise and is now dependent on the frequency of the input signal, see the graph in figure 7.19. Therefore, for a given input signal $x(t)$ and sample rate (f_s) there will be an optimum step size (k') that will follow the curve given by the graph in figure

7.12. If a range of input signal frequencies is to be used, then the optimum step size (k') must be chosen for the maximum input frequency.

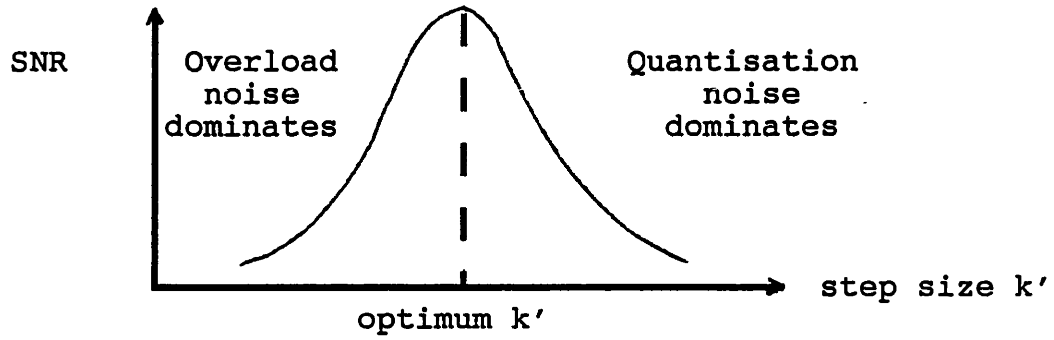


Figure 7.12. Optimum step size.

An analysis of the slope overload noise effect can be made more quantitative by considering, as an example, a sine wave test signal. Say,

$$x(t) = A \sin wt.$$

The maximum slope of this signal is $dx/dt = Aw$, therefore the DM should be able ideally to respond to this slope. If not, then the ramp of the modulator output will deviate from the sine wave at some point before the zero crossing. See figure 7.13.

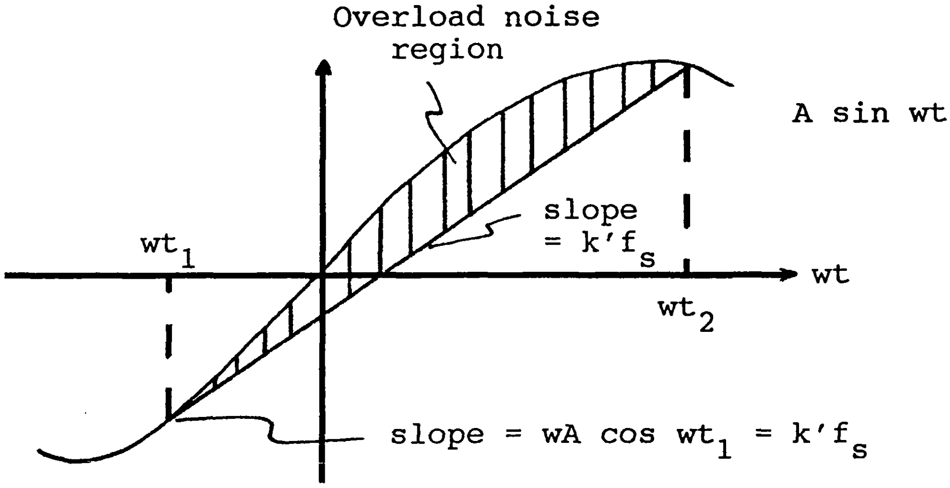


Figure 7.13. Noise caused by slope overload.

The portion of noise due to overload (N_o) can be obtained from the average of the difference between the true signal voltage and the modulator output voltage.

$$N_o = \int_{-wt_1}^{wt_2} \left(A \sin wt - \left(k'f_s\left(\frac{wt+wt_1}{w}\right) - A \sin wt_1 \right)\right) dwt$$

substituting $\alpha = \frac{k'f_s}{wA}$

$$N_o = A \int_{-wt_1}^{wt_2} (\sin wt - \alpha(wt + wt_1) + \sin wt_1)\, dwt$$

$$N_o = A \left[-\cos wt - 2\,\alpha\,(wt)^2 - \alpha\, wt_1 + \sin wt_1 \right]_{-wt_1}^{wt_2}$$

$$= -A \cos wt_2 - 2\,\alpha\, A\, (wt_2)^2 + A \cos -wt_1 + 2\,\alpha\, A\, (wt_1)^2$$

$$= A \cos wt_1 - A \cos wt_2 + 2\,\alpha\, A\, ((wt_1)^2 - (wt_2)^2)$$

$$= A\ (-2\sin(w(t_1 + t_2)/2)\sin(w(t_1-t_2)/2) + 2\alpha w^2(t_1^{\ 2}-t_2^{\ 2}))$$

If we now assume $t_2 = 2t_1$

$$N_o = 2A\ \left(\sin \frac{3wt_1}{2} \sin \frac{wt_1}{2} - 3\ \alpha\ w^2\ t_1^{\ 2} \right)$$

The point of slope overload is given by

$$wt_1 = \cos^{-1} \alpha$$

so that

$$N_o = 2A\ \left(\sin \left(\frac{3 \cos^{-1} \alpha}{2}\right) \sin \left(\frac{\cos^{-1} \alpha}{2}\right) \right)$$

Therefore to avoid slope overload $\alpha \geq 1$. Thus the delta-modulator must be able to follow a slope of Aw to avoid overload noise, but quantisation and slope overload noise are additive and therefore,

$$SNR = 20 \log \left(A / \sqrt{(No^2+Nq^2)} \right)$$

As an example, the derived equations can be used to illustrate the noise performance of a delta-modulator, using $A/k' = 500$ and $f_s = 300$ kbps, the DM follows the curve shown in figure 7.14.

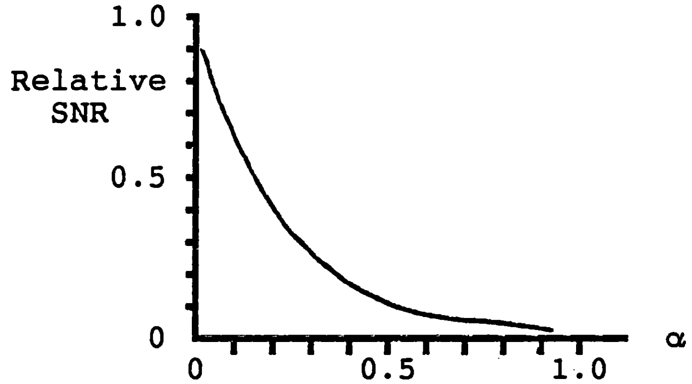


Figure 7.14. Response of a delta-modulator

A comparison of coding techniques is shown in the graphs in figures 7.18-7.20 and clearly, the simpler version of DM is vastly inferior to PCM systems at all but the lowest bit rates. The improvement offered by CVSD is to reduce the slope overload effect (see figure 7.15) and therefore the sampling rate can be reduced. The adaptive step size means that the element of noise caused by slope overload will be much smaller.

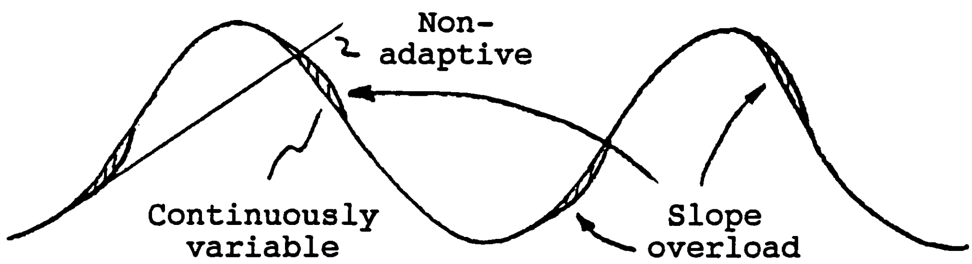


Figure 7.15. Noise caused by a CVSD modulator.

Motorola [23] pages 9-14 and 9-20 have already calculated SNR for their CVSD CODEC algorithm. The curves for CVSD on the graph in figure 7.18 were obtained from this reference.

7.2.2 Experimental Results

An assessment of signal quality implies an assessment of fidelity. However, word score tests involving human perception and even articulation index methods tend to measure a lack of fidelity rather than the opposite effect.

The coding techniques being investigated here can all be made to perform to an acceptable level of fidelity given the right set of circumstances. Note that VOCODER techniques are excluded because they tend to have poor speaker recognition and they also have considerable variability in quality with a range of speakers in different acoustic environments [48].

Partly to prove the technical concepts of combining multiple simultaneous audio inputs, but mainly to evaluate the relative speech quality of various digitizing techniques, a circuit was built using the TMS32010 digital signal processor (DSP), which could combine three separate inputs, independently, adjust their individual gains, scale and round, and present the resultant to an operator headset whilst hosting algorithms for different coding techniques.

A prediction of the speech quality performance is in subsequent sections of this chapter, followed by a review of relevant subjective tests. Later an objective method of observation will be discussed. The test sentences used in subjective tests were provided to the Tactical Information Division at GEC Sensors by the Royal Aircraft Establishment (RAE) at Farnborough. Some of the most useful sentences, containing awkward words and phrases, are included in appendix II.

In a laboratory environment, the following observations were made:

CVSD	@	16 kbps	poor quality
ADPCM	@	16 kbps	fair quality
CVSD	@	32 kbps	good quality
ADPCM	@	32 kbps	good quality
CVSD	@	90 kbps	very good quality
A-law PCM	@	96 kbps	very good quality

These observations relate to how comfortable the audio output from a coding technique was to listen to as well as the scores awarded between talker and listener with random word tests.

Subjectively, 90 kbps CVSD and 96 kbps A-law PCM were similar in performance and of an order required for an aircraft CCS. Therefore these two techniques required further investigation.

The graphs in figures 7.21-7.25 were obtained from measurements performed on the evaluation circuits. They show the characteristics of CVSD and A-law PCM at various bit-rates and at different input signal amplitudes. The graphs display bandwidth and distortion for the given variables, leading to the conclusion that little difference exists between the two schemes for a given bit-rate, where that bit-rate is of the order of 96 kbps.

7.3 Speech Quality Index

Schemes have been devised [25-27] to make the measurement of SNR in communication channels more methodical and repeatable. These schemes have varying degrees of success when used to evaluate different digitizing techniques. Steeneken [26] has already made a partial evaluation of digital techniques using a SNR index method but this only looks at the effect of bit errors. ie. channel noise rather than coding noise. The use of optical fibres or some other high integrity transmission medium can greatly reduce channel noise and therefore coding noise is the dominent effect.

7.3.1 Articulation Index

Articulation index (AI) as devised by French and Steinberg in 1947 was a method for predicting speech intelligibility of a communication channel from its physical parameters. The method was reconsidered by Krypter in 1962 who greatly increased its accessibility by the introduction of a calculation scheme. Measurement of the index has more recently been made automatic by using microprocessor based circuits to perform the calculations [27].

The calculation of AI basically consists of three steps:

(a) The calculation of SNR within a number of frequency bands. The original AI scheme was based on 20 equal audio-frequency power bands, between 250 and 7000Hz. For reasons of simplicity, a reduction to five octave bands was made in later AI schemes.

(b) A linear transformation of the effective SNR to an octave-band-specific contribution towards the index, ranging from zero to one.

(c) The calculation of the weighted mean of the contributions of all relevant octave bands constitutes the AI.

The final index may take a value between zero and one where zero indicates an unintelligible scheme and one indicates a fully intelligible scheme.

7.3.2 Rapid Speech Transmission Index

The AI method is particularly appropriate for channels with distortion in the frequency domain such as interfering noise and bandpass limiting, but it is not as applicable when non-linear distortions such as peak clipping and distortions in the time domain such as the slope overload of delta-modulation are involved.

The method suggested by Steeneken and Houtgast [25], although based on the AI concept, is more generally applicable as it also accounts for disturbances in the time domain and for non linear distortions. They have called their method the speech transmission index (STI). This technique has been used in a piece of equipment marketed by Bruel and Kjaer in Holland and they call the measurement the rapid speech transmission index (RASTI). The STI method, like the AI method, has three elements. They are as follows:

(1) The channel index test is treated phenomenologically [28]. The SNR per octave band is determined by means of measurements performed with a special test signal [26].

(2) Due to the specific nature of the test signal, reflecting the spectral and temporal characteristics of running speech, the method also accounts correctly for non linear distortions.

(3) A new dimension is introduced to evaluate distortions in the time domain. Here the fluctuation rates encountered in running speech are considered. This is achieved by the inclusion of an extra modulation component in the speech test signal.

The STI is ideally suited to the measurement of such effects as reverberation and echoes, the main advantage being the nature of the special test signal. This test signal has the same frequency spectrum as the average long-term spectrum of speech. Together with an arbitrary spectrum of interfering noise, a sinusoidal intensity modulation with well defined modulation index is applied to the test signal. The SNR at the receiving end of the channel can be derived from the observed decrease of the modulation index. Distortions in the time domain essentially affect the envelope of the signal. Hence for each octave band the modulation index has to be measured as a function of modulation frequency, resulting in the modulation transfer function which gives a representation of the transmission channel.

As stated earlier, Steenken and Houtgast [26] applied their STI to some digital speech systems to investigate their performance across poor channels (high error rate). The STI or RASTI system is a rigorous analysis which provides a single index to predict the transmission quality of a communication channel. Although the method can be used as a base to continue research, the requirement in this chapter is an analysis that will provide an indication as to which digital speech coding technique performs best for a range of input frequencies and amplitudes, and over a range of serial bit rates.

7.3.3 Digital Speech Quality index (DSQI)

A new index can be devised to give specific information about different audio coding schemes. The variables against which the quality measurement in this index are made are frequency, amplitude, bit-rate and background noise. This creates a four dimensional model or rather a graph with five axis if these parameters are to be monitored. The analyses shown in section 2.1 of this chapter take these variables into account and shows how some techniques perform as these parameters vary. What is required to appraise the digitization techniques is an index which will vary when any one of the four parameters is varied. This method lends itself to computer techniques, especially modern graphical representation, providing a three dimensional picture equivalant to those used in computer aided aeronautical design, making it easier to determine suitable and relative performance criteria. The production of a computer graphics generating program is outside the scope of this thesis. However, a program to generate a matrix of graphs or tables (see figure 7.16) based on the preceding description would be feasible if an absolute indication of which technique to use for a given set of conditions is required.

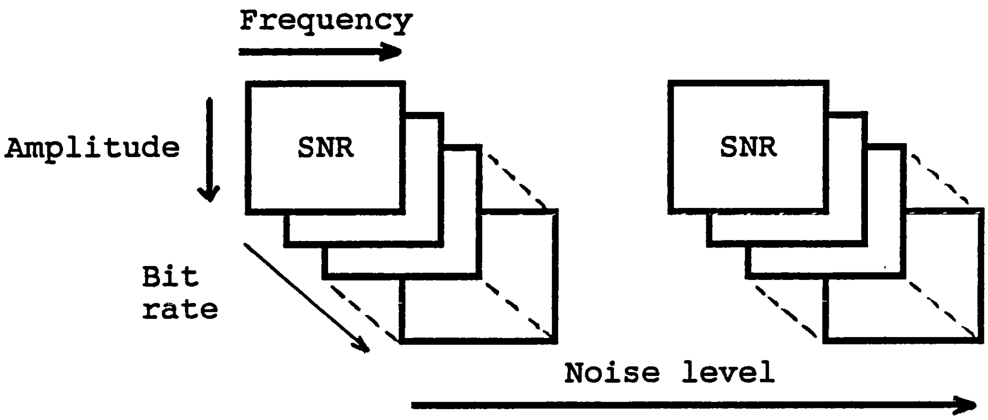


Figure 7.16. DSQI analysis.

7.4 Calculating the Effect of Bit Error Rate

Another factor to play an important role in the perceived quality of a digital audio distribution system, is the error performance of the transmission path.

The number of random errors on the data passing around the network can be quantified as the bit error rate (BER). For fibre optic links this is expected to be of the order of one error every thousand million bits [12].

There are a number of conflicting arguments that have an effect on the choice of scheme to digitally encode speech, one of which is an investigation to establish the significance of the position of errors within a packet (a packet is shown in figure 7.17). The object here is to establish whether data errors were significant compared with header errors.

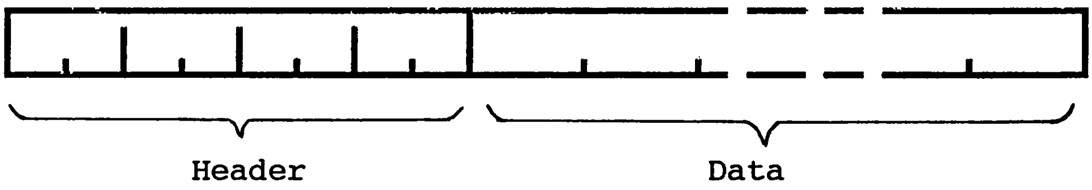


Figure 7.17. Packet Format.

With 32 bits per header and 224 bits per packet, the probability of an error being a header is,

$$P_{eh} = 32/224 = 0.1429 \qquad (1)$$

and the probability of it being a data field error is,

$$P_{ed} = 1 - P_{eh} = 0.8571 \qquad (2).$$

The significance of error location in a packet is that an error in the header will cause the loss of a whole packet whereas a data error will cause the loss of either one sample or a single bit depending on the coding scheme.

If one packet represents one period of timed speech (T_p), then one sample (8 bits for A-law PCM) represents 0.04167 of a period and one bit (CVSD) represents 0.005208 of a period, then normalized to one packet, with a network bit error rate P_e, the length of speech lost because of a header error is,

$$T_h = (P_e P_{eh}) T_p = 0.1429 P_e \qquad (3)$$

the length of speech lost because of a sample error is,

$$T_s = (P_e P_{ed}) 0.04167 = 0.03574 P_e \qquad (4)$$

or the length of speech lost because of a bit error is,

$$T_b = (P_e P_{ed}) 0.005208 = 0.004464 P_e \qquad (5).$$

The effect of a sample data error against a header error is,

$$100 T_s / (T_h + T_s) = 20\% \qquad (6)$$

and the effect of a bit data error against a header error is,

$$100 T_b / (T_h + T_b) = 3\% \qquad (7).$$

The values in (6) and (7) show that, although PCM errors can be more significant than CVSD errors (especially MSB), a bit error in the header field is far more serious than a bit error in the data field. These figures lead to the conclusion that to increase reliability, error checking should be adopted for the header field but is not necessary for the data field. If a high BER transmission channel were used, then the evidence would have to be reconsidered.

Putting these figures into the perspective of time; for a data rate of 10^7 bits per second and a bit error rate of 10^9 bits per error, at each of the total number of nodes (N_C) that may cause an error, a rate of one error every $100/N_C$ seconds will be expected to occur on the ring. However, some errors will occur in the gaps between packets and all errors will be shared between all 80 nodes on the network (if less than 80 nodes are connected, the difference can be classified as empty packets). This leads to one error per node, every

$$\frac{100}{N_C} \times \frac{256}{224} \times 80 = \frac{9143}{N_C} \text{ seconds}$$

Therefore, for an 80 node ring, with 8-bit samples, in 25600 seconds, there will be 32 lots of 2ms glitches (header) and 192 lots of 83µs glitches (sample). For 1-bit samples there will still be 32 lots of 2ms glitches but the 192 lots will be of 10μs glitches, in the same period of time. This equates to approximately 263ms of errors per day for PCM coded systems and 220ms of errors per day for delta-modulation systems. Again, the percentage difference is not insignificant. But over such a long period of time, the difference will be perceptually negligible.

7.5 Summing Up

To accommodate a higher processing capacity and perform some other networking functions, the more powerful and recently available Motorola 56000 DSP can be used along with CODEC IC's. This can optimise complexity and performance trade-offs.

Objective performance measures are not well established and are typically used only as a relative comparison in CODEC design. Formal judgements on coded speech quality almost inevitably depend on subjective testing. Seveniratne [11] page 105, states that quality figures (eg. AI) only give a comparison between systems and subjective tests are still needed to measure adequate quality. The best subjective test for comparison of high quality speech coding systems is the diagnostic acceptability measure [54]. Other test sentence material is also useful, including isopreference tests [53] and modified rhyme tests [55].

All of the coding techniques considered, score fairly high on intelligibility (a CCS must have reasonably high quality so as not to degrade the performance of equipment connected to it). Some however, are easier to listen to than others. The techniques that perform most favourably for the requirements of this thesis are 90 kbps CVSD and 96 kbps A-law PCM.

Curves in [26] show that a comparison of AI methods to perceived quality at the higher end of the quality measurement become less reliable. This must always be considered when working with index measurements at the higher quality end of the scale.

The graphs and measurements shown throughout this chapter show a comparable quality for 90 kbps CVSD and 96 kbps A-law PCM and both are possible solutions.

The effects of channel errors discussed in section 4 of this chapter have some bearing if a low quality link (such as a twisted-pair) were the requirement. In this case a CVSD scheme would be preferable because of its superior performance over PCM in the presence of transmission channel errors.

Although 90 kbps CVSD and 96 kbps A-law PCM provide comparable quality performance, the circuit complexity and signal processing must be considered when aiming at a marketable product. The preferred choice is A-law PCM because it is easier and quicker to process than a CVSD signal.

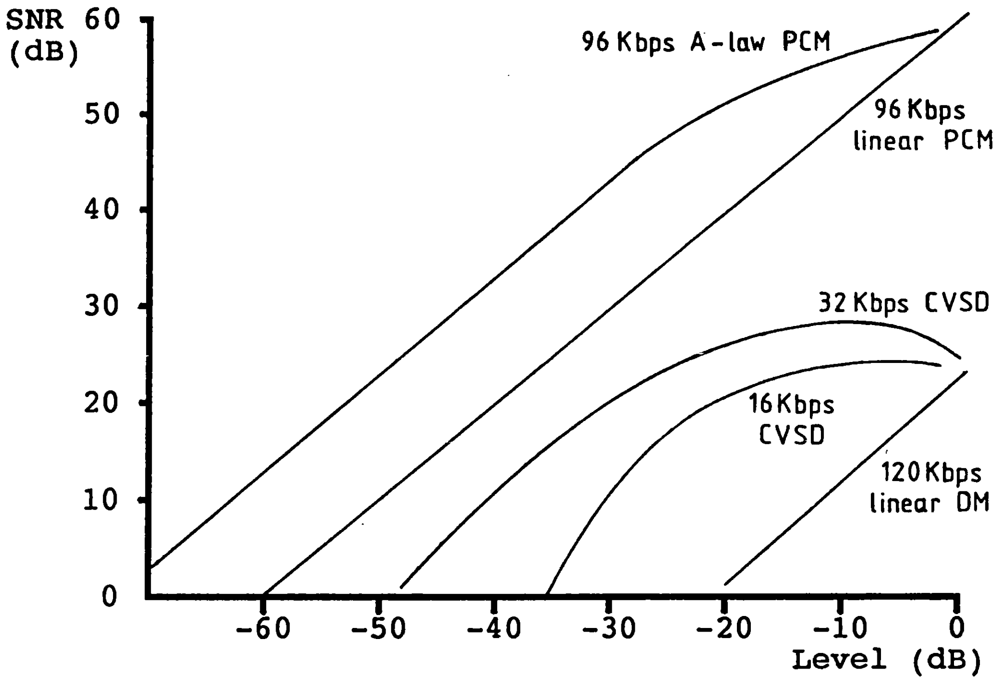


Figure 7.18. Relationship between SNR and level with respect to maximum input amplitude for a 5 kHz audio system.

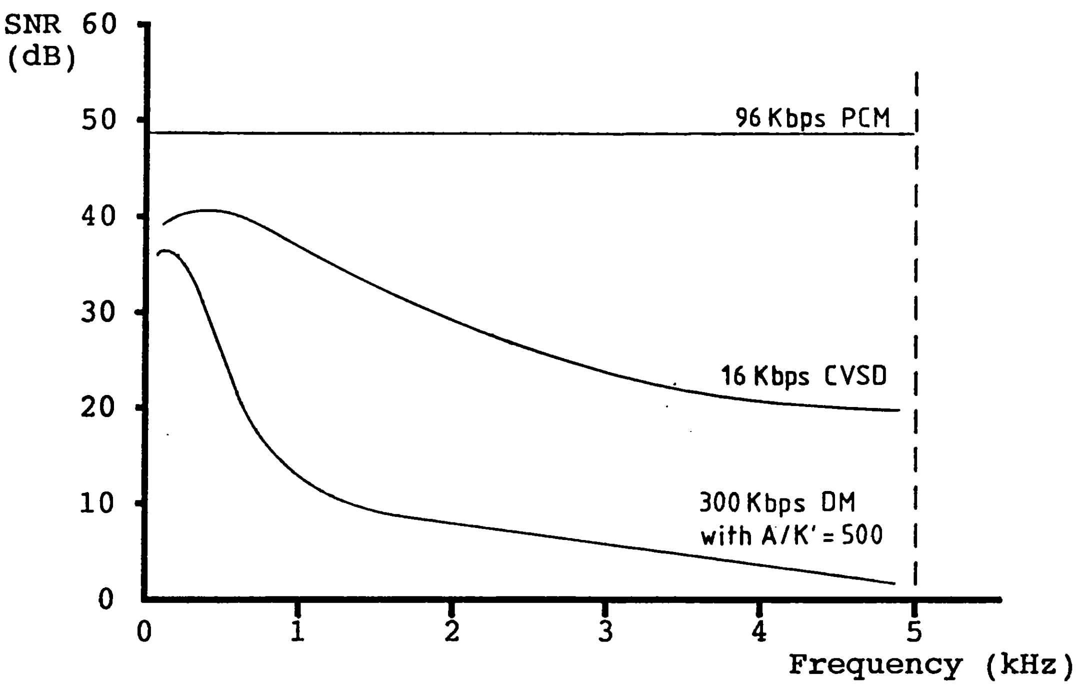


Figure 7.19. Relationship between SNR and input frequency for a 5 kHz audio system.

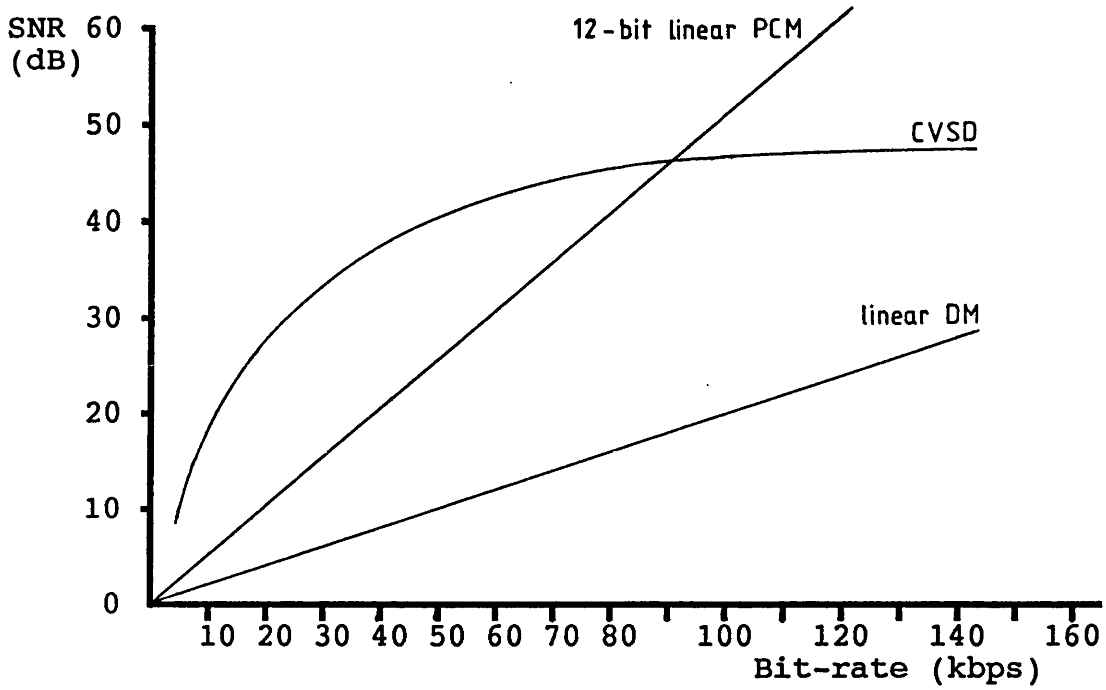


Figure 7.20. Relationship between SNR and bit-rate for a 5 kHz audio system.

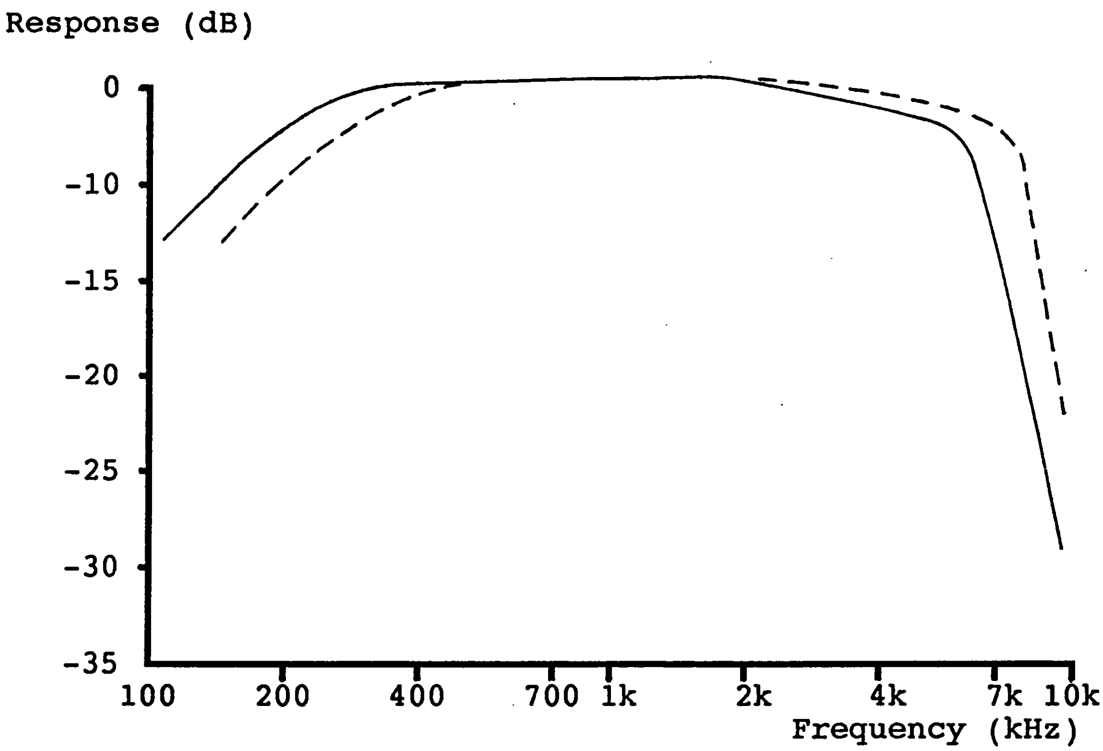


Figure 7.21. Bandwidths of CVSD and A-law PCM, at the same serial bit-rate, for a 5kHz audio system.

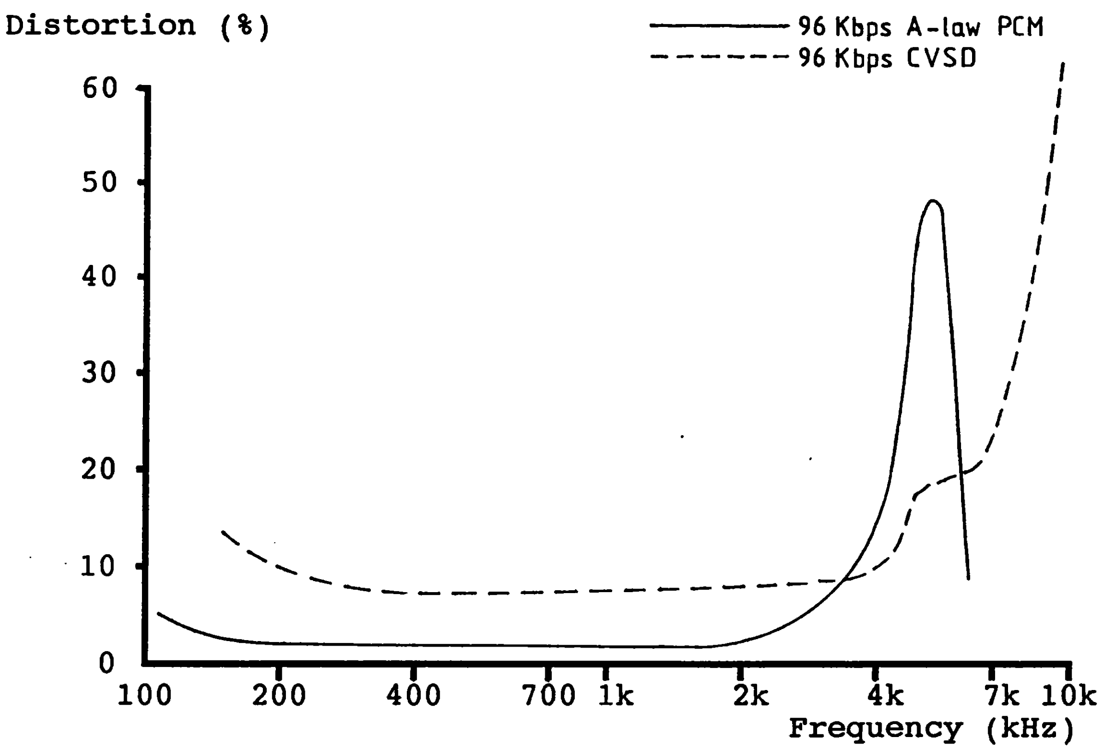


Figure 7.22. Distortion of CVSD and A-law PCM at the same serial bit-rate.

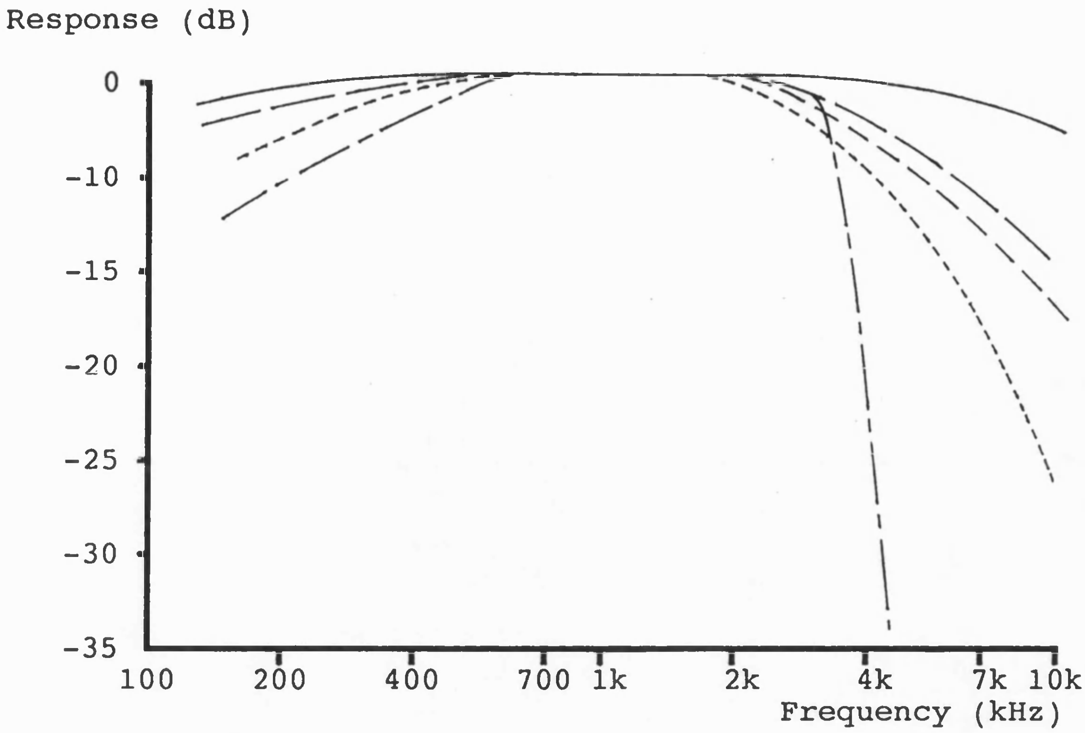


Figure 7.23. Filtered bandwidths of CVSD and A-law PCM, at different bit-rates.

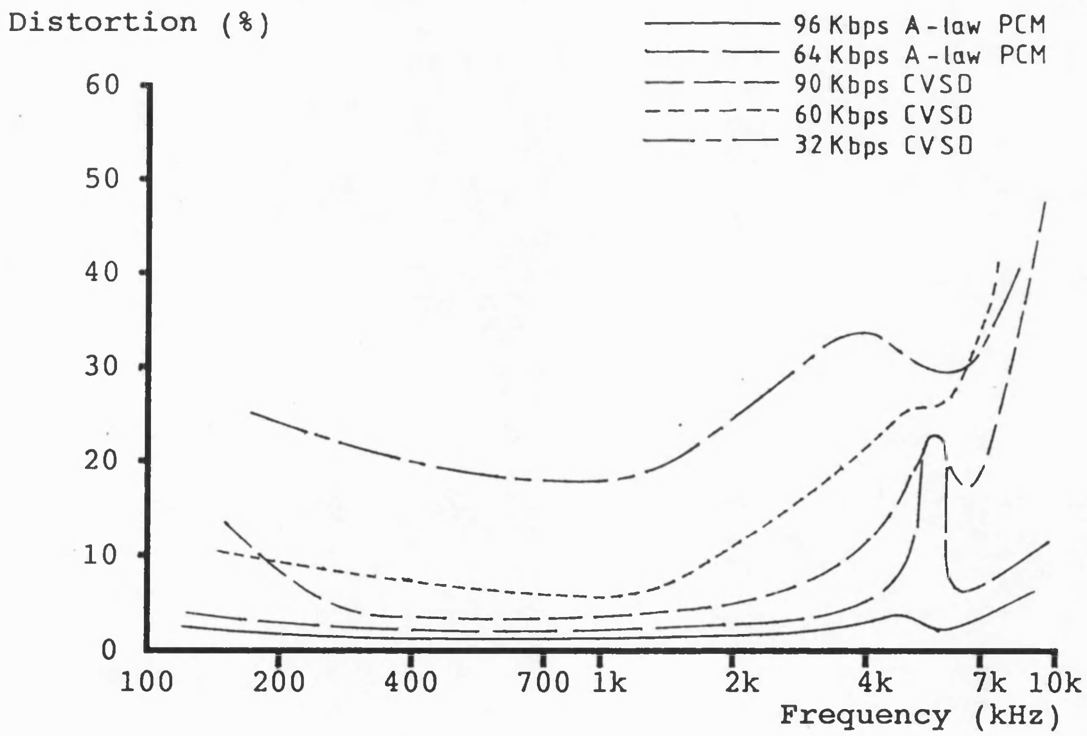


Figure 7.24. Distortion for CVSD and A-law PCM at different bit-rates.

8. PRACTICAL SOLUTIONS

Having established in chapters 6 and 7 the principles by which the system is intended to operate, the design of a practical solution is needed so that features which need clarification may be proved.

The independent gain adjustment and summation of analogue signals in conventional centralized aircraft CCS has been achieved using multiple operational and trans-conductance amplifier circuits and was implemented in large central units. The target of this implementation is to perform the same functions while the audio signals are still in digital form and to distribute those functions throughout all the nodes connected to the network. This will require digital multiplication and accumulation.

Therefore, for simultaneous channel reception it was necessary to build and investigate an experimental system with a minimum of three independent transmission stations with at least one of them being capable of reception as well. Using this number of stations provides for evaluating side-tone and at least two speakers with one listener, see figure 8.1. This mini-system (nicknamed the Speech Processing Evaluator) is independent of network protocol and architecture, which could have introduced further uncertainties to audio quality measurements, and was used to prove sufficient intelligibility could be maintained.

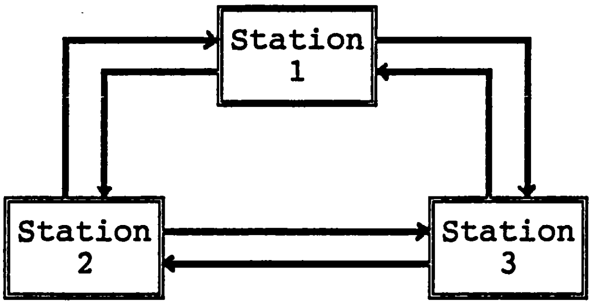


Figure 8.1. Speech Processing Evaluator interconnections.

This practical implementation attempts to incorporate features of existing microprocessor designs and uses a combination of hardware and software techniques.

This chapter will discuss in some detail the circuits, software and evaluation before describing the working demonstration.

8.1 Speech Processing Evaluator (SPE)

An initial survey of ways of performing the digital multiplication and accumulation showed that if a 12-bit analogue to digital converter (A/D) and a 12 kHz sampling rate (for reasons already explained in previous chapters) are used, then a 16-bit microprocessor is preferable for speed and convenience. At this stage a purely hardware solution was discounted as it would have not only produced a bulky expensive board, but would have lacked versatility. Figure 8.2 shows a block diagram of one station of the SPE.

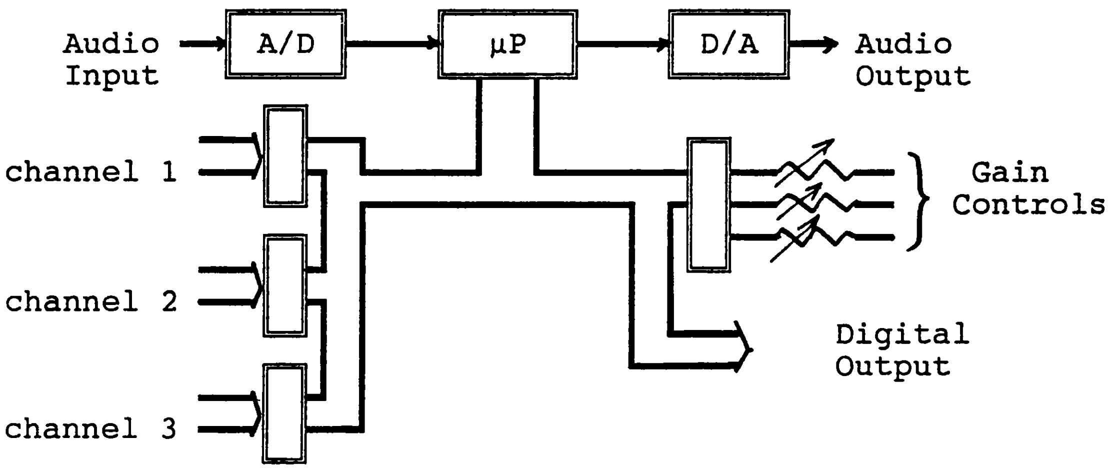


Figure 8.1. Block diagram of intended solution

The microprocessor needs to perform the following functions every 83 μs (1/12000 second):

SPEECH INPUT CYCLE:-

Sample : store / convert : transmit.

SPEECH OUTPUT CYCLE:-

Receive : convert : multiply / accumulate n times : scale and round : de-sample.

where n is the number of channels to be listened to simultaneously and "convert" is the optional process of converting linear PCM to another form of code.

8.1.1 The choice of microprocessor

The speech output cycle listed above is similar in operation to a digital implementation of a finite impulse response (FIR) filter. The difference is that the SPE will not be taking fixed coefficients from memory locations. These are replaced by the gain coefficients and are multiplied with concurrent rather than sequential samples. The choice of microprocessor depends upon the ability to perform these functions sufficiently fast.

An Intel 8086 can perform a single tap of an FIR filter (1 multiply / accumulate) in approximately 2.1 milliseconds. This time would be extended further in the SPE because of inputting new gain coefficients and concurrent samples. Using an 8087 arithmetic co-processor with the 8086 reduces the required time to 27 microseconds. This is still not fast enough because the function must be performed several times within a single 83 µs period. A much faster solution is the Texas TMS320 digital signal processor (DSP) which takes only 200 nanoseconds for a multiply and accumulate instruction.

Using a bit-slice processor would have been faster than the TMS320 but is a less elegant solution for the same reasons as the hardware solution.

The use of a DSP provides the further advantage that different speech coding schemes can be tested for their relative performance with software and minor wiring changes to the circuit board.

It is important to remember that sampling and de-sampling follow strict timing because of the periodic nature of speech samples. Failure to comply with this rule will result in unwanted distortion. The sample output cycle is a good example of where this may be a problem. A latch must be provided between the microprocessor and the D/A to ensure that the length of processor instructions does not affect the absolute sample time. A copy of an SPE station circuit diagram is included in appendix III.

8.1.2 Circuit considerations

The detailed design of the microprocessor circuit is not a relevant aspect of this thesis. However, Texas Instruments (TI) suggested the use of non-erasable bipolar PROMs with their TMS32010, because no UV erasable EPROMs were available to interface to the processor with sufficiently high access speeds.

This is an expensive way of developing programs because each PROM can only be used once. To make experiments and tests easier to perform, a method of down-loading the program from slow EPROM into fast RAM was incorporated. This can be seen in the circuit diagram in appendix III.

The circuitry for the SPE was designed on an IBM PC/AT work-station using the Futurenet, Dash-4 circuit drawing tool.

8.1.3 The SPE mini-system

Once the SPE circuits, utilising the TMS320, had been designed (see appendix III) and constructed it was possible to test them. A program was written which would generate a ramp output (simply a case of incrementing the output value), and then, having verified the processor circuits, the analogue circuits were proved by writing a program that could output a sine-wave. The sine-wave table used was generated from a BASIC program (see appendix V) on a VAX computer, to avoid manual entry of instantaneous sine-wave amplitude values. The table was incorporated into the program and the successive values were routed to the digital to analogue converter (D/A).

The analogue circuitry was designed to interface to a NATO standard headset. The headset provides a nominal microphone signal of 1.5 millivolts (rms) and requires a nominal output to the ear-piece of 2.7 volts (rms), an increase of 6 dB to allow for maximum amplitudes, gives 5.4 volts, so the circuitry must accommodate a peak-to-peak signal of $5.4 \times 2\sqrt{2} = 15.27$ volts. Therefore either power rails in excess of this value or a step-up coupling transformer are required.

To prevent values resulting from multiplication and addition exceeding the maximum range of the components, scaling and rounding must be incorporated into the software. The scaling is used when two numbers are multiplied. This is achieved by shifting the resultant value to the right. The rounding function is performed after the required number of additions. If the resultant exceeds the maximum for a 12-bit number it is rounded to hexidecimal FFF. Likewise if it is negative, the value will be rounded to 000. Figure 8.3 shows how the digital output amplitude might vary for a given signal.

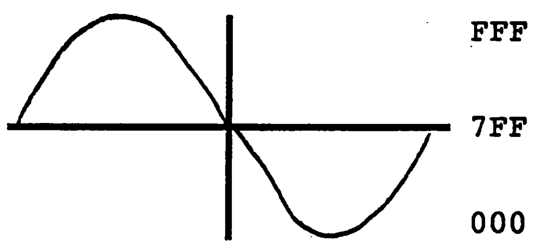


Figure 8.3. Hexadecimal representation for a sine wave.

The A-law companding (compression) matrix mentioned in chapter 7 was generated by a BASIC program, then loaded into an EPROM from a VAX computer. A copy of both the BASIC program and the generated table are included in appendix V.

Having provided an A-law compression table for coding microphone signals, the equivalant expansion table is also required. This calls for a further BASIC program on the VAX (appendix V), the table being merged with the main program (see appendix IV).

The capacity of the SPE for simultaneous reception of multiple conversations, when coding speech for A-law companding is 12 channels. This number was obtained by measuring the time taken to process one channel and dividing it into the 83µs available. For some applications this may not be enough, therefore, in a production system a faster DSP will be required.

The SPE was built then evaluated using a number of test subjects from within the Tactical Information Division (TID) at GEC Sensors. The tests were based on recordings made within TID, on cassettes, using a hi-fi recorder. Both male and female speakers were recorded, reading rhyme words and structured sentences supplied by the Royal Aircraft Establishment (RAE) at Farnborough. Shaped noise was also provided by RAE in the form of aircraft recordings on reel-to-reel tape, so that the recordings of speakers could be played through the system with varying amounts of background noise. The audio output of this system was presented to various listeners for their relative opinions. It was observed that both CVSD and A-law PCM could support a sufficient audio quality.

8.2 Implementation

Having shown with the SPE that the simultaneous reception of multiple conversations could be supported digitally with sufficient audio quality, the next stage was to produce a mini-system that was capable of demonstrating the local area network protocol as well as all those features from the SPE.

One particular feature of the LAN that cannot be completely tested without a full implementation is the clock. Although clock reconstitution can be proved, the phase jitter that is usually expected to occur on a large ring with a multiplicity of nodes must be avoided. Although this effect can be reduced by a method already discussed in chapter 6, phase jitter can be minimised if a crystal controlled phase-locked-loop (PLL) is used.

This has the advantage that the frequency will not drift very far in the event of the clock not being locked. The requirement is locked frequency but not locked phase.

The frequency of the incoming data can be divided by the appropriate integers to give a range of possibilities for voice coding standards, thus falling in line with the model of chapter 4 based on the 7-layer approach. An example of this is shown in figure 8.4, the important feature now being the interface between layers of the design. The particular feature shown in figure 8.4 shows a seemingly un-characteristic transfer of an element (clock information) from layer 1 (physical) to layer 6 (presentation). It will therefore be necessary to observe at all times, that this element is available at all levels and is theoretically passed between all layers. A more detailed account of the 7-layer approach follows in the next section.

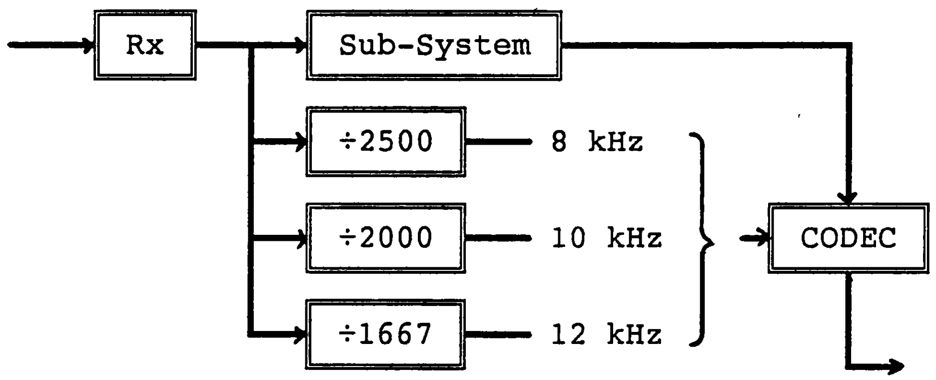


Figure 8.4. Locking sample frequency to network clock.

Despite IC manufacturers being unusually pessimistic about their CODEC chips, samples have been found to perform satisfactorily beyond their specified performance. The Mostek MK5326 was shown to work at a sampling rate of 16 kHz instead of the 8 kHz specified, before any non-linearity was observed. Use of CODECs is a progression from the speech processing evaluator (SPE) as the availability of CODECs provides a worthwhile reduction in component count.

Packets need to be sorted into order, therefore a novel organisation of buffer memory is required. Each new packet arriving at a receiving node contains 24 sequential samples (see chapter 3). Each subsequent voice packet in a 2ms period that is decoded by that node, contains the same number of samples that, for a given epoch, conceivably occurred at the same instant as for the other packets from different sources, see figure 8.5. This being the case, a matrix memory format is required such that samples may be entered one packet at a time and removed in groups of coincident samples. This process also necessitates the use of a fast implementation as several samples must be removed every 83 microseconds (1/12000 second) and packets may be input at a rate of 10 Mbps (to a limit of approximately 20 packets in a 2 ms epoch).

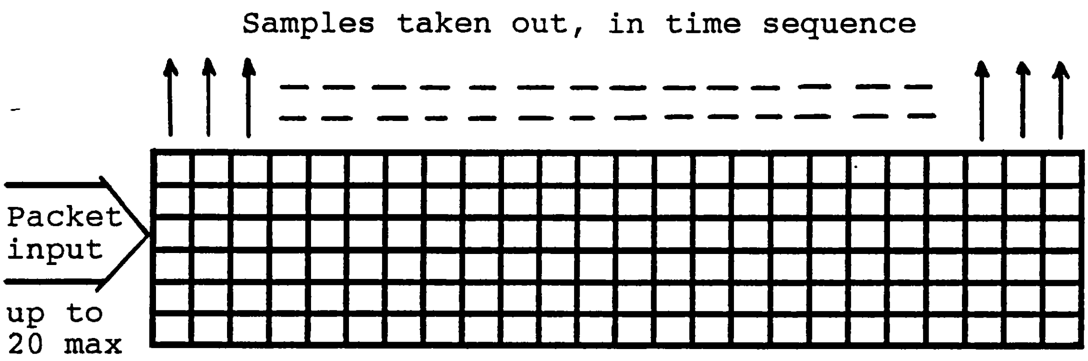


Figure 8.5. Matrix memory format.

This memory matrix could be built from dual-ported memory devices with counters and multiplexers, but a smaller more cost-effective solution was developed and will be described in section 8.3.1.

8.2.1 7-layers

The advantages of arranging the design as an open-system to make use of the 7-layer model [13] have already been listed in chapter 4.

Important features when applying the 7-layer model to an implementation are information passing between software layers and handshaking across hardware interfaces. Figures 8.6 and 8.7 show the contents of the 7-layers for this implementation.

LAYERS	SPEECH-OUTPUT	SPEECH-INPUT
7) APPLICATION	CODEC applies signal to operator audio interface.	Filtered audio into CODEC.
6) PRESENTATION	Data presented to CODEC at serial interface.	Data passed from CODEC into microprocessor.
5) SESSION	Data extracted from matrix memory in sample format.	Samples put in queue.
4) TRANSPORT	Data transferred into matrix memory according to packet header.	Header is added. Voice/data interleaved.
3) NETWORK	Address recognition	Synchronsation of packet output.
2) DATA-LINK	Header check Parity check	Add parity Wait for empty slot.
1) PHYSICAL	Optical to elect. conversion. Manchester decode. Clock recovery.	Manchester coding Elect. to optical conversion.

Figure 8.6. Layered approach to voice processing.

LAYERS	CONTROL PANEL DISPLAY ↑	CONTROL PANEL INTEROGATION ↓
7) APPLICATION	Pass information to serial interface.	Data passed serially to interface.
6) PRESENTATION	Decide on reaction.	Evaluation of requirements.
5) SESSION	Compare with data-base.	Compile message.
4) TRANSPORT	Type of message, (BITI, reconfigure, or radio)	Add supernumerary information to message.
3) NETWORK	Address recognition	Synchronisation of packet output.
2) DATA-LINK	Header check Parity check	Add parity Wait for empty slot.
1) PHYSICAL	Optical to elect. conversion. Manchester decode. Clock recovery.	Manchester coding Elect. to optical conversion.

Figure 8.7. Layered approach to data processing.

The application of the 7-layer model in this case is a contrived version of the standard [13]. Its use with voice data rather than for computer applications means that a variant of its implementation is necessary.

It can be seen from figures 8.6 and 8.7 that the lower three levels (layers 1-3) are identical for both voice and data applications. From layer 4 upwards, specific functions take over.

As data leaves the fibre optic ring network at 20 Mbps, it is passed serially through layer 1, to layer 2. The packets, having had their clock regenerated in layer 1, are clocked in a constant stream into a shift register in layer 2 with a 3.2 μs gap every 25.6 μs. No hardware hand-shaking is provided between these two layers, so that any data not transferred is lost.

Data passing from layer 2 to layer 3 is queued in byte form. Hardware interrupts enable layer 3 to read from layer 2 when the queue is the correct size. At this stage the packet header has already been checked for synchronisation and parity.

When the packets are ready to pass to layer 4, unwanted ones are rerouted back to the network. Packets that have been originated from within the node in question have their empty bit reset, and all packets are passed to layer 4 at the output of the gate array and also back to the network output. In layer 4 control data and voice data are treated differently.

In the opposite direction data passes from control or voice sub-systems of layer 4 onto layer 3 in the form of a queue. Clock synchronisation exists between the two layers such that data is not lost. Extra data is added in layer 3 so that data is passed to layer 2 in the same way as the previous layer.

Packets are passed from layer 2 to layer 1 as 10 Mbps serial data.

Features of the layered approach that are specific to the voice sub-system are as follows:

In the network to earpiece direction, the connection is fixed by a microprocessor peripheral interface.

Between layers 5 and 6 data is handled by two totally independent interrupt routines, therefore, the matrix memory format is arranged so that changes in either routine are accommodated so long as they comply with matrix format.

In the opposite direction, 8-bit samples are offered from layer 7 to layer 6 every 83 μs using a clock synchronised with layer 6. The samples are queued into layer 5 where the header is added to a packet within an 83 μs period, then the whole packet is passed to layer 3.

For control data, information passed between layers 4 and 5 takes the form of request packets as described in chapter 6. From layer 5 to layer 6 appropriate subroutines are called depending on the particular request. This operation is similar in both directions. Layer 6 to layer 7 transfers contain data which is to be displayed to an operator.

In the opposite direction, when a key is pressed on a control panel, data is passed from layer 7 to layer 6 where it is stored as a number of bytes in memory. In layer 5, data is formatted into 24 byte packets and an extra 4 bytes of header are added in layer 4 ready for packet transmission onto layer 3.

Figures 8.6 and 8.7 are for an operator interface node. Radio and other audio connection nodes will be similar.

8.3 Demonstrator

The demonstrator being designed and built to prove the operation of distributing digitised voice on a local area network was conceived and implemented onto a single printed circuit board known as the "Core-Card" (appendix I). This helps to achieve a compact node design which can be repeated at several positions throughout an aircraft. The demonstrator consists of enough nodes to demonstrate multiple intercom and radio management facilities.

The implementation of the demonstrator falls into three areas; firstly voice processing, covered by the digital signal processor (DSP), secondly medium access control (MAC), for the distribution of data throughout the network, and thirdly the control area, consisting mainly of software, and used to ensure the operational success of the system.

8.3.1 DSP

It has already been established that a new generation of DSP must be used in the demonstrator. The choice of available devices widened considerably in the time between the SPE design and that of the demonstrator. One device receiving wide publicity during this period was the Inmos Transputer. The Transputer was not chosen because it lacked the "on-board" multiplier associated with other DSPs. The most suitable candidates were:

Analogue Devices	ADSP2100
Texas Instruments	TMS32020
Motorola	DSP56000

The Motorola DSP56000 was chosen because although being equal in most aspects to the other processors, it has a CODEC interface and the added capability of performing very fast interrupts, so making it suitable for the direct memory access (DMA) function required for the matrix memory element.

Advances in memory devices matched those in DSPs so that 55 ns access time EPROMs are obtainable and can be directly accessed from the DSP unlike the program down-load method incorporated into the SPE.

Calculations for the maximum number of simultaneous channels that can be processed by the DSP56000 yield 19, showing, as expected, that it is superior to the TMS32010 which only manages 12.

With respect to figure 8.6, the DSP56000 is used to cover layers 4-6 on the speech output cycle and layer 6 on the speech input cycle.

8.3.2 Medium Access Control

The medium access control (MAC) part of this system equates to layers 1 to 3 of the reference model described in chapter 4. The MAC implementation is achieved using programmable gate arrays. These gate arrays were designed using the XILINX computer-aided-design tool on an IBM AT work-station. Their operation was simulated on the computer to test that the circuits perform as required. The iterative process of simulation and re-design proved the circuit design to be sound.

The gate arrays contain the functions of address recognition, voice/data packet split, parity check, slot access, and other functions that have already been described in chapter 6.

The actual connection between node and network medium requires a conversion between electrical and optical signals in the most modular form. Fibre optic transducer modules are used for convenience. These modules are state-of-the-art at present being capable of transmission at 50 Mbps with LEDs and including clock reconstitution. It is thought that for a production system, discrete fibre optic amplifiers will be used as these can be made cheaper and just as small as readily available modules. A further reason is that manufacturers have no plans to militarize their modules in the near future.

The gate array portion of the MAC corresponds to layers 2 and 3 of the model, with respect to both figures 8.6 and 8.7. Layer 1 corresponds to the physical interconnection, bit-rates, clock recovery etc. Note that the speed at which these gate arrays operate is very relevant. If fast enough gate arrays had not been available, a solution incorporating ECL would have been required.

8.4 Software

Several management type functions still remain in each node, not being accommodated by either the DSP or the gate-arrays. These functions are defined with layers 4-7 in figure 8.7 only. For these functions a microcontroller was used.

It is this microcontroller that contains software which is used to make decisions on how to use the self-healing mechanism and operate the switches which form part of the contra-rotating ring.

The software in the microcontroller also performs the function of constructing all control packets, for both automatic radio management and reporting built in test information (BITI).

The final function of the microcontroller is to manage the interface with the operator's control panel. For the purpose of the demonstrator, BBC Master micro-computers are used as operator control panels. This function also provides a path between the operator and the DSP for the volume control data.

8.5 Summary

The SPE was used for all digital speech tests. It proved (subjectively) that the desired speech quality could be maintained whilst transmitting speech between nodes, processing the digitized speech and adjusting the audio amplitude of the various component signals. A key feature to be tested on the SPE was the transmission of speech in the presence of noise. This was achieved using recordings of male and female speakers superimposed on varying levels of noise that had been recorded directly from an aircraft. The predominant from test subjects throughout these tests, was that as the noise level increased, any imperfections in digitized speech quality became less noticeable.

The 7-layer reference model approach to system design has proved invaluable. Without this tool, the passing of information and the transfer of data within each node would have become an awesome task to unravel.

9. INTEGRITY AND RELIABILITY

Integrity and reliability are such important issues when considering aircraft systems that they warrant a chapter of their own. The requirement to produce a high integrity system necessitates a reliable ring design as well as a reliable top level system design.

The first level of operational confirmation for the Digital Communications Control System (DCCS) in normal operation is for the operator to hear audio through his headset. At the very minimum, when the operator speaks, on hearing his own voice (side tone), he has proved the operation of the network and can be sure that all other connections to the network have been offered his microphone signals. Subsequent levels of built-in-test (BIT) accommodate practices that have become universal when designing for modern embedded microprocessor applications.

System reliability depends in part on the failure modes, and the redundancy that is used to protect against failures and damage. The failure modes fall into two categories; those associated with signal failures in the electronics - node failures; and those associated with disruption of the optical signal in the fibre - bus failures. The first of these two categories can be caused by either component failures or by power failure, whilst the fibre failure can only be caused by breakage or inadvertent disconnection.

Redundancy can be added to virtually any level desired but each new level incurs additional weight and cost penalties. At the highest level, radios and signal routing are paralleled in aircraft communication systems to provide a flight essential safety feature.

9.1 Ring Integrity

The fibre optic ring used to realise the method of information distribution proposed in this thesis must be a very reliable element in the system. The nature of a ring as opposed to a bus dictates that certain extra precautions need to be taken.

It may appear initially that rings are inherently less reliable than bus networks. In practice however, the MTBF of rings with a multiplicity of nodes can be very large, and good diagnostics can provide very low MTTR figures. A high MTBF combined with a low MTTR combine to keep a very high availability figure.

9.1.1 Reliable rings

All nodes connected to the ring are permananantly in the data path and some mechanism of establishing that failures do not cause the system to cease functioning must be incorporated. There are a number of enhancements that can be made to ensure a high availability.

One approach is the "braided-ring". Each node is connected to a successor and to that nodes successor. The network can be gracefully degraded within limits, as components fail. However, this technique does rely on the braids being sound , and because they are only used under fault conditions, they may have faults themselves which only come to light when they are needed.

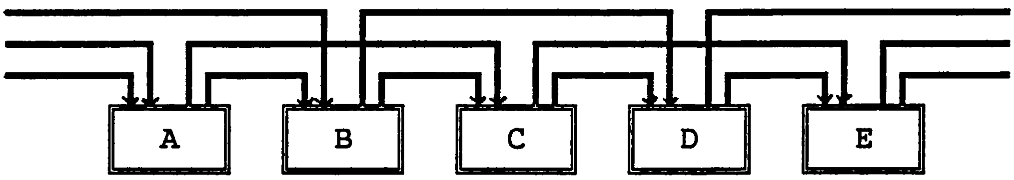


Figure 9.1. Braided-ring interconnections

Another approach is the "witch centre". The ring is formed into a star shape with reconfiguration capability at the centre. In the event of failures, a given spur can be switched out. This scheme has the disadvantage of introducing a central element which may be prone to battle damage in a military environment. In any environment, failures in the central unit may cause the whole system to fail.

A choice now exists between the use of a dual-redundant uni-directional ring and a self-healing contra-rotating ring. These will now be examined and compared.

A single uni-directional ring has the possibility that a single node failure could block all data flow past that node. This is unacceptable in a high integrity aircraft environment as the MTBF of the ring is equal to the MTBF of one node.

The dual redundant uni-directional ring also has the possibility that a single node failure could block all data flow past that node, but fibre, connector and local access failures can be bypassed by the redundant ring. This does not compare favourably with the contra-rotating ring (see figure 9.2), which has a minimum MTBF of twice that of one node. The minimum ring failure occurs when two nodes have failed (assuming node failures to be more probable than fibre failures) causing the main ring to split into two smaller ones. However, if the nodes were adjacent, or the excluded nodes were non-essential, the failure would be less than catastrophic.

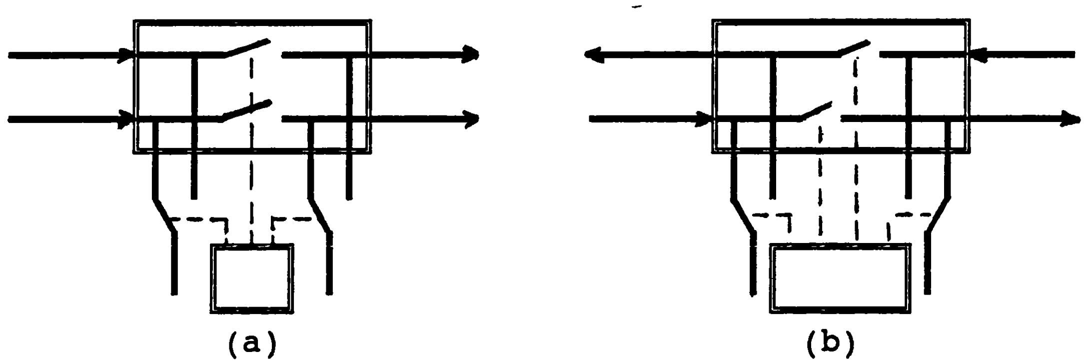


Figure 9.2. Ring reliability.

(a) Uni-directional and (b) contra-rotating

The reliability of the ring network can be greatly improved by the careful design of loopback and bypass techniques at each node to ensure that the reliability and failure characteristics are compatible with high integrity avionic requirements. The flow chart of a design to meet these requirements is shown in figure 6.12 (page 96) and covered by Russell [10].

The operation of the self-healing switches on the ring, must not interrupt speech when re-routing the data path. As this is an aesthetic issue rather than a reliability one, the performance of the self-healing ring will be considered in chapter 10 rather than here. Suffice to say that their operation is not of sufficient length to prevent a "may-day" call from getting through.

The switch configuration of the contra-rotating ring proposed here is versatile enough to allow for a further mode of operation whereby the second ring can be used in preference to the first ring. Although it is not intended for use in this implementation, it is a further element of redundancy and can be incorporated if required.

9.1.2 Fibre Optic Reliability

Using optical fibres with point to point links, excellent noise immunity, electro magnetic pulse (EMP) immunity and lack of earthing related problems can be obtained. Optical fibres also improve electro magnetic compatibility (EMC) and TEMPEST performance. TEMPEST is previously defined in section 3.5.3 (page 21).

The number of random errors on the data passing around the network can be quantified as the bit error rate (BER). For fibre optic links recommended by Harrison [18], this is expected to be around one error every thousand million bits (10^{-9}).

Other factors affecting fibre optic reliability include mechanical aspects, life of connectors, ability of fibres to withstand vibration, bending for prolonged periods and termination failures. However, these will depend on the aircraft in which they are implemented and therefore they will not be considered here.

9.1.3 Optical transmitter reliability

With ageing, LEDs exhibit gradual degradation in terms of optical power output. This degradation occurs because of heating effects when in use. This may be as a result of crystal defects, impurity migration and for extremely high quality devices, as a result of recombination enhanced point defect generation.

It is the crystal defect induced degradation which is of concern to a fibre optic ring network [45]. The non-radiative current increases, and the output optical power decreases accordingly, appearing as dark lines in the emitting area, giving rise to the term dark line defect (DLD).

In terms of degradation, the useful life of an LED is determined by the system in which it operates. If, for example, the optical link performance is such that when the transmitter optical output power falls by 30%, insufficient power is incident on the receiver, then the useful life is different to that for a link where a fall of 70% can be tolerated. For the purpose of these calculations, a fall to half (3dB) of the original value will be used.

The temperature and current dependence of the degradation process is described by Arrhenius's law:

$$\tau = \tau_0 \exp(E_a/kT)$$

where τ_0 is current dependent and assumed to have power-law relationship [45] and depends on the device being used. The following abbreviations are used:-

τ = degradation characteristic time constant (MTBF in hours)

k_1 = rationalisation constant

I = operating current (mA)

E_a = activation energy (0.458 eV)

k = Boltzmann's constant

T = temperature (K) and

n = -2 for surface emitting devices

Substituting for τ_0 with the parameters for the Hewlett Packard HFE4000 series devices [45], the mean time between failures can be shown as

$$MTBF = \tau = k_1 \ I^n \ \exp(E_a/kT).$$

From figures supplied by Hewlett Packard for their LED emitters, with $k_1 = 157$, a continuous drive current of 40 mA and running at a temperature of 70 °C, the figure for MTBF is 530 x 10^3 hours (60 years). However, degradation only occurs when the device is switched-on, therefore, for a 50% duty cycle code such as Manchester, the MTBF is doubled to 10^6 hours (120 years).

For an application where high reliability and minimum short-term failures are critical, increased service life can be assured by screening units subjected to burn-in and removing those which exhibit excessive short-term power output degradation rates. Burn-in methods are empirical with effectiveness increasing with longer burn-in periods, higher burn in temperatures and tighter limits on acceptable power output degradation over the burn-in period; all of these with corresponding increasing costs.

Now that it has been established that LED failure is by degradation rather than catastrophic, it follows that this degradation can be monitored for the purpose of maintainance reporting. Once the power level at the receiver has been detected to drop to a pre-determined level (the ring is still functioning normally), the node down-stream from the degraded emitter can be made to transmit this information using BITI. This information can then be used by the maintainance crew. The effect of optical receiver degradation is less significant [45].

9.1.4 Power Supply Distribution

The in-line nature of the optical-to-electrical and electrical-to-optical conversion elements means that for the ring to continue normal operation, these must not only contain very reliable components but must also continue their operation without interruption. As the converters are active components, they require a power supply which must also be reliable.

Large-aircraft power distribution takes several forms. As with automobile electrics, some equipment can be connected directly to the batteries (nominally 28 volts) and the rest of the equipment is connected to alternators (115 volts at 400 Hz). Aircraft systems have a third option not available on cars which is the 28 volts derived from the alternator for charging the batteries. Aircraft supplies are also split into three categories; vital, essential and non-essential. The vital supplies are connected directly to the batteries (through fuses) and therefore the current consumption must be kept to a minimum. With this in mind it is desirable to connect only the electrical/optical conversion elements to this supply. This is however, an area of work which will vary with each application to a new aircraft.

The object of making reliable power distribution links to each node may be eased if the power cables are installed along side the fibre optic cables. Reliability is increased because if the power supply cable is severed, it is likely that the optical fibre has also been severed. Partial reliability improvement is obtained here as optical connections may still be maintained on the other side of the node.

9.2 System Monitoring

To simplify the maintainance and repair of this system, a degree of monitoring is used. The main object of this monitoring is to report fault and performance information to operator control panels on request, and where fitted, to a communications services monitor (CSM). Relevant built-in-test information (BITI) can also be provided to the ground crew for maintainance.

9.2.1 BITI

Built-in-test (BIT) is a feature of most modern microprocessor based circuits enabling several of the elements in the design to be monitored for failure or degradation.

A watch-dog timer can be used to ensure that if for some reason the microprocessor stops using its buses, the timer will reset the node. Continuing failures of this nature can be used to select the ring bypass switches.

A power monitor can also be provided to operate in the same way as the watch dog timer.

The microprocessors can be used to periodically check their memory. This is less likely to cause a catastrophic failure and more likely to cause degradation of the service at that node because a microprocessor failure will not necessarily cause a node to fail completely.

Non-catastrophic failures will be reported to the CSM or a control panel. Catastrophic failures however, are bypassed if detected, or removed from the ring by loop-back if undetected at the faulty node.

9.2.2 Packet removal

The operation of the ring whilst data is flowing is a potential area for errors. If a data packet should be corrupted by circuit errors or by transmission line errors (BER $= 10^{-9}$), there must be a means for removing that damaged packet.

Packet removal is normally performed at the originating node, once the packet has made a full cycle of the ring. This is possible because each node checks source addresses. If the source address has been corrupted, the originating node is no longer able to remove its own packets. It will be the responsibility of the operating monitor node to remove that packet.

The operation of monitor mode is based on a single-bit flag and the initiation of this facility was described in chapter 6. It functions as follows:-

If empty bit is set, pass through.

If full and monitor bits are set, reset both.

If full bit is set, then set monitor bit.

This procedure ensures that no packet exceeds two cycles of the ring. If packets were to continue to circulate, this would have an adverse effect on data bandwidth by filling some of the circulating slots with disowned packets, leaving less for the good packets.

9.2.3 Virtual Path Integrity

Virtual path links (all the circuitry between the two ends of a communication path) between operators and radios, and between operators and encryptors are also an area requiring some discussion from the system reliability view-point.

The links are set-up by a request, response and confirmation operation. Therefore, failures in the set-up procedure are overcome when the procedure is incomplete, by restarting the procedure. The option of making the virtual path into a higher integrity link exists by initiating regular confirmation messages in addition to the normal voice packets. On a failure to comply with a confirmation message, relevant information can be stored in BIT form, the link is de-selected, and the node at the other end of the communication path can be informed.

9.3 Fault Indication

As a result of compiling BITI and any other information that might be required, a communication path for reporting to the CSM or a control panel must be established.

Although the information that is presented to operators and maintainance crew does not really contribute towards the improved reliability of the system, it does aid the rapid location and isolation of faults and make their repair a quicker operation. Hence, enhancing reliability through BIT induced maintainance. This reduced repair time means that multi-million pound aircraft have to spend less time on the ground and can spend more time in the air and hence availability is increased. It is also worth looking at being able to transmit BITI to the ground via a radio link so that in the future, repair time may be reduced even further.

9.4 Summary

The requirement to implement the Russell Ring into a high integrity aircraft environment has dictated a system with good reliability. This chapter has highlighted those features of the system that are designed to improve reliability, and also those features which need to be considered for each new application in different aircraft.

Use of fibre optics, whilst improving noise immunity, EMC etc, has introduced new reliability problems that have been considered.

Overall system integrity has been shown to improve with BITI and fault indication.

10. PERFORMANCE EVALUATION

It is at this stage in the thesis that a comparison of achievements with requirements can be made. The requirements that can be compared for performance are;

- expandability,
- simultaneous reception of multiple signals,
- "open microphone" inputs,
- single failure tolerance,
- minimal degradation with multiple failures, and
- integrated voice and data.

First, the suitability of the ring to meet the target size and specification will be considered Then some further investigation will follow to show how the ring network performs beyond its design requirements.

10.1 Analysis

The expandability of the system up to a size of 80 nodes is an inherent feature of the design. This can be illustrated quite clearly. Each node that needs to transmit continuously (up to 80) requires to have access to the ring every 2 milliseconds (the amount of speech in each packet). As long as this access is obtainable, no information will be lost because of size boundaries.

The application of an "open-mic" (node continuously transmitting data onto ring), means that the access to the ring will depend on the number of nodes that are transmitting (N_t) at any one time. The mean access time (μ) will be given by:-

$$\mu = \frac{25.6 \times N_t!}{(N_t + 1)^t}$$

where 25.6 microseconds is the average time period of one packet plus one gap. The effect of the number of nodes connected (N_c) to the ring is not relevant because N_t nodes are equally likely to be contiguous in a small N_c system as they are in a large N_c system.

The central tendency of the distribution of access to the ring, about the mean, will vary according to the the total number of nodes (N_c) connected to the ring. This is because the number of available empty slots will increase as the total number of nodes increases. To illustrate this, an 80 node ring with 5 nodes transmitting will have a higher probability of accessing an empty slot quickly than a 20 node ring with 5 nodes transmitting.

The following table shows the mean access delay times against the number of nodes with "open-microphones".

Number of nodes transmitting	Mean access time (μs)	Maximum access time (μs)
1	12.8	25.6
2	25.6	51.2
3	38.4	76.8
4	51.2	102.4
5	64	128
10	128	256
20	256	512
30	384	768
50	640	1280
78	998	1996.8
80	-	2048

Figure 10.1. Table of ring access times.

This shows, as expected, that the mean access delay increases depending on the number of nodes transmitting, for a given number of nodes connected.

Note that with 80 nodes transmitting the maximum access delay exceeds the 2 milliseconds required for no information loss. This is considered reasonable as with 80 nodes transmitting on an 80 node ring, no one will be listening anyway. The point at which this is no-longer a nonsense scenario is not clearly defineable.

The requirement for the system to work with sizes varying from approximately 10 nodes connected, to a maximum of 80 nodes, is easily satisfied with a ring bit rate of 20 Mbps. It must be remembered that audio signal performance is the eventual subject for discussion, and the numbers here all contribute to illustrating that performance. Although these calculations have some bearing on audio quality, that subject has already been discussed in previous chapters.

10.1.1 Control Panel Response

The second system function to have a direct effect on the operator is the response time of his control panel (D_{cp}). However, an absolute value cannot be evaluated here because a portion of the time will depend on the particular control panel used. The contribution added by the ring can be evaluated.

Constituents of the delay (D_{cp}) are:

> The time to compile a message (T_{cm}). This is the time taken between the press of a button on the control panel and a packet being ready to transmit. For each node this corresponds to layers 3-7 of the reference model as the control panel is to be excluded from the calculations.

The mean time spent waiting for an empty slot (μ). This depends on the number of nodes transmitting at a particular time.

Transmission delay (D_t), including electrical to optical conversion, fibre delay and optical to electrical conversion for a full cycle of the ring.

Time taken to store the message (T_{sm}). From packet received, to message understood (a software feature).

Compilation of a response (T_{cr}). The reaction of the receiving node to produce a packet that is ready to transmit (a software feature).

Then four further processes of:

Wait for an empty slot,
transmission delay,
store message and
transmit to control panel.

Note that there will be a further delay dependent on the total number of nodes on the ring. This is the case because the combined request and response time takes one complete cycle of the ring.

Therefore, the response time can be obtained from;

$$D_{cp} = T_{cm} + 2\mu + 2T_{sm} + T_{cr} + N_t(D_t + 25.6)$$

where T_{cm} and T_{sm} are $<10\mu s$, T_{cr} is $<100\mu s$, μ can be obtained from the table in figure 10.1 and D_t is approximately 20ns. The following can now be obtained;

for $N_t = 20$, $D_{cp} < 1.25ms$ and

for $N_t = 80$, $D_{cp} < 5ms$.

These figures compare favourably with conventional systems in which response times could sometimes be half a second.

10.2 Probabilistic Performance

A more rigorous examination of network performance shows that it performs adequately within its design constraints.

10.2.1 Queueing Theory

The analogy can be drawn with conventional queueing theory [46] where a packet (or sample) is equivalent to a customer arriving at a server, which now becomes the receiving node. The service time can be thought of as the time interval between a packet arriving at the node, and being stored. This queueing and serving takes the form of first-in-first-out (FIFO). If however, the end of service time (departure) is taken from when the sample leaves the memory to be processed by the DSP, the operation would be service-in-random-order (SIRO). The former case will be used for this analysis.

Using Kendall's nomenclature from Cooper [46] chapter 5,

$$a \,/\, b \,/\, c : d \,/\, e \,/\, f$$

where

a = arrivals distribution

b = service time (or departures) distribution

c = number of parallel servers ($c = 1 - \infty$)

d = service discipline (eg. FIFO, LIFO, SIRO etc.)

e = maximum size of system (queue + service)

f = size of calling source,

and distributions can take the form,

M = Poisson (Markov) arrival or departure distribution

D = deterministic inter-arrival or service time

E_k= Erlangian (gamma) distribution of inter-arrival or service time

G_i= general independent distribution of arrivals

G = general distribution of departures (or service time).

Queueing at nodes can therefore be displayed as,

$$G_i \ / \ D \ / \ 1 : F \ / \ 20 \ / \ 80$$

Knowing this information, a number of general observations can be made. Firstly, the steady state probability of having n packets in the node during a time (t), assuming equal probability for n = 0 to 19, is;

$$P_n(t) = 1/20 = 0.05$$

A maximum of 20 is chosen as an operator is unlikely to need more simultaneous audio channels selected. If t is 2ms, one epoch, then there is an equal probability that the number of packets accepted by the node will be between 0 and 19 (the mode and mean will depend on operational scenarios and crew positions). Therefore, assuming the average case and taking a mean of 10 packets being accepted in one 2ms epoch, gives a mean arrival rate α = 5000 packets per second. The expected size of the system (L_s) will be,

$$L_s = \Sigma\ nP_n = 9.9$$

and the mean queue length (L_q) will be,

$$L_q = \Sigma\ (n-c)P_n = 8.55$$

where c is the number of parallel servers. The expected waiting time in the system (W_s) is,

$$W_s = L_s/\alpha = 1.98\text{ms}$$

and the expected time for waiting in the queue (W_q) is,

$$W_q = L_q/\alpha = 1.71\text{ms}.$$

This shows that under peak operating conditions, each node will be functioning within its design capabilities.

10.2.2 Expansion Capabilities

There may be a requirement in the future, and indeed the system would be more marketable, if it was possible to expand it beyond the normal size of 80 nodes. To this end, the possible effects of using the network with more than 80 nodes will be examined.

It has already been shown that as the number of transmitting nodes increases above the design limits, system performance will be degraded. If either bit rate is reduced or the number of nodes increased, the effect of losing packets will be probabilistic. If an empty packet cannot be accessed within 2 milliseconds, it will be scrapped and hence lost.

The three examples that will be considered here are an 80 node ring operating at 5 Mbps, a 255 node ring operating at 20 Mbps and a 200 node ring operating at 50 Mbps.

The graph in figure 10.2 is based on the probability (P_{lp}) of losing a packet because of insufficient empty packets being given by;

$$P_{lp} = 1 - \frac{\text{CAPACITY}}{N_t} \qquad \text{for } N_t \geq \text{CAPACITY}$$

Where the capacity is the maximum number of nodes that can be accommodated for a given bit rate once control packet requirements have been included.

Capacity = 20 nodes for a 5 Mbps ring.

Capacity = 80 nodes for a 20 Mbps ring.

Capacity = 200 nodes for a 50 Mbps ring.

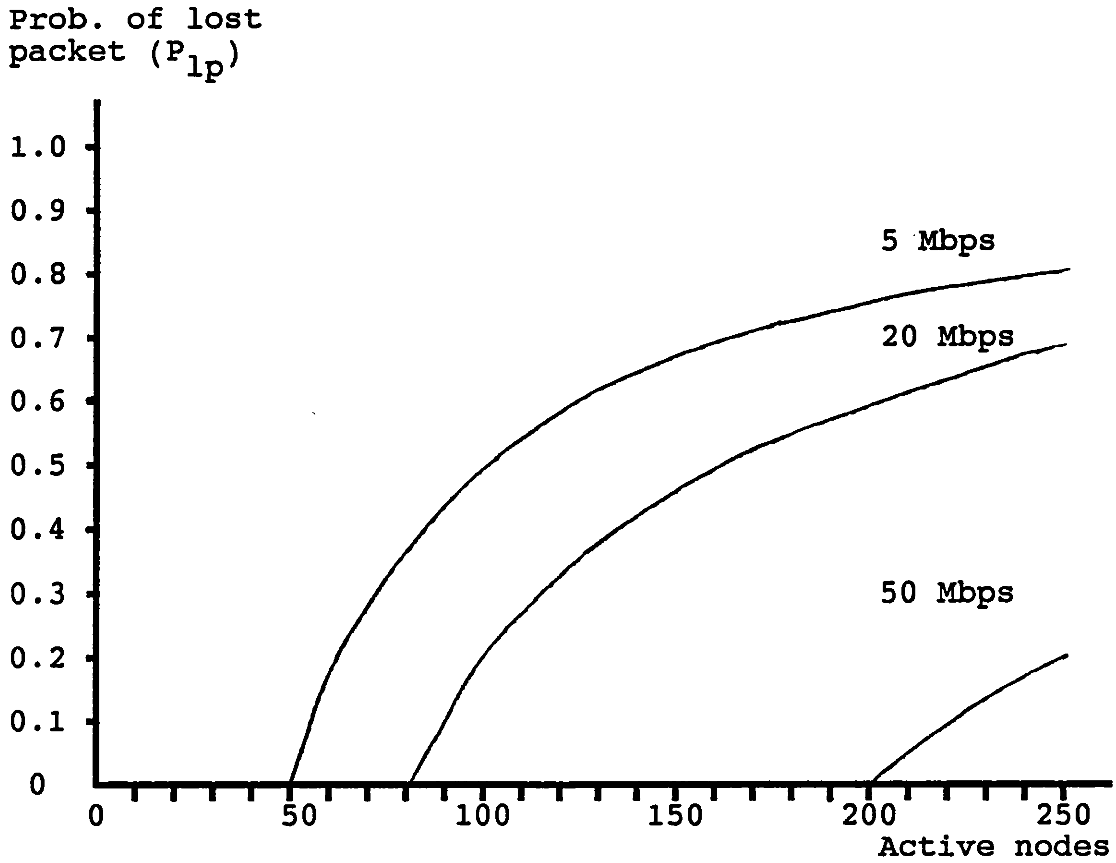


Figure 10.2. Graph of node access probability.

The probability shown in figure 10.2 is related to the distortion that will be observed at the subsequent audio output. This type of distortion has already been quantified subjectively in chapter 2, section 3.2, where burst errors accounting for 6.4% of lost speech samples was barely acceptable. For each node, this is equivalant to loosing a 2ms speech packet every 31.25ms. A ring with a capacity of 80 nodes will lose a packet every 390µs to obtain the same level of distortion for all nodes. The mean access time for nodes on a 20 Mbps ring is 25.6µs. Therefore, using these numbers in a modified form of the packet loss formula gives,

$$N_t = \frac{CAPACITY}{1 - P_{1p}} = 85$$

The maximum number of transmitting nodes on a 20 Mbps ring will be of the order of 85 to avoid unacceptable distortion. However, the example in chapter 2 used 16 kbps CVSD to illustrate the point. This already has a higher level of distortion than 96 kbps A-law PCM which would be expected to perform better.

To continue looking at the effect of losing packets, the probability of one channel losing one packet is as shown in the graph in figure 10.2. The probability of two channels losing one packet each, will be half the graphical figure, as will the probability of one channel losing two packets (P_{12p}). $P_{12p} = P_{1p}^2$ such that using the original formulae for lost packet probability would yield a much smaller figure for the number of nodes transmitting continuously (N_t) before two sequential packets are lost hence causing the same level of distortion.

10.3 Comparison with other LANs

The efficiency advantage of using the ring proposed in this thesis, for the particular applications being considered may be seen in the throughput graphs in figure 10.3 [35]. They show that by sending packets with fixed length and fixed access boundaries, the response of data throughput compared to bit rate is linear. If short bursts of information are transmitted, the efficiency is high throughout the whole traffic loading range. However, when large bursts are transmitted, this is achieved in the proposed system by sending several small packets. The efficiency is then less than some other protocols as the overheads form a more significant proportion of the overall packet length.

To provide a full, large packet facility, one of the extra cascaded packets could be used for error checking and correction, and for overall packet size indication. The new level of efficiency would be given by,

$$E = \frac{\text{number of bytes required to be transmitted}}{\text{number of bytes that must be transmitted}}$$

$$= \frac{24n}{28n + 672}$$

where n is the number of bytes to be transmitted, 24 is the number of data bytes in each packet and 28 is the total number of bytes in each packet.

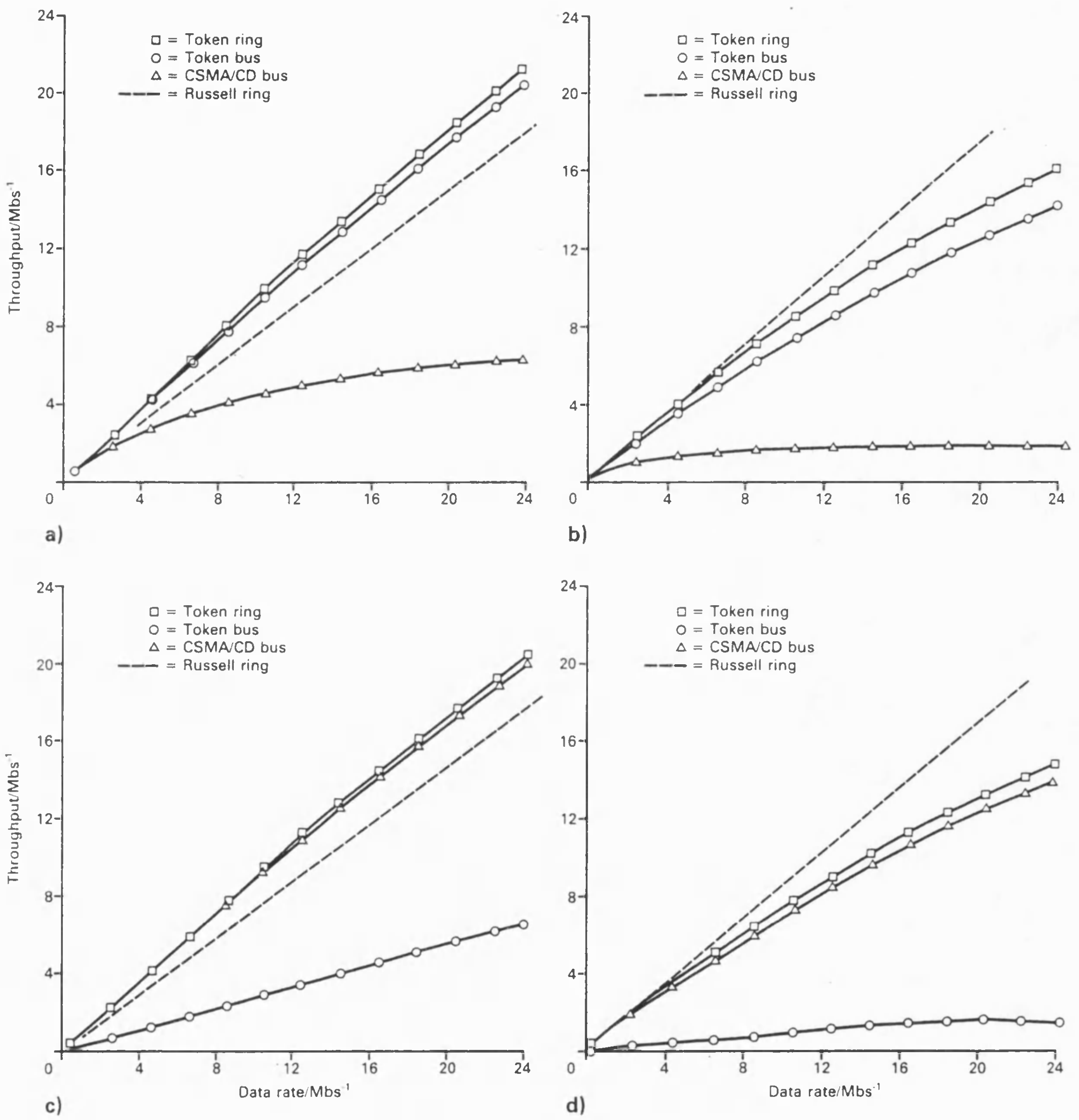


Figure 10.3. a) 2000 bit packet, 100 stations active b) 500 bit packet, 100 stations active c) 2000 bit packet, 1 station active d) 500 bit packet, 1 station active

Removing packets at their destination rather than waiting for them to return to their source would have the effect of doubling the throughput, but in the case of intercom, packets have multiple destinations and therefore this practice is impossible.

10.4 Clock slippage and epoch errors

The successful operation of the ring is in part dependent on the accuracy of crystal oscillators. In this section it will be shown that the errors caused by clock tolerance will not be significant.

The frequency differences found in crystals may cause epoch boundaries to move. This in turn leads to bunching or spreading of packets, both resulting in the introduction of errors. In figure 10.4, these could be seen as the movement in time of leading and trailing packets relative to an epoch boundary. A question to be answered now is, how often will this occur?

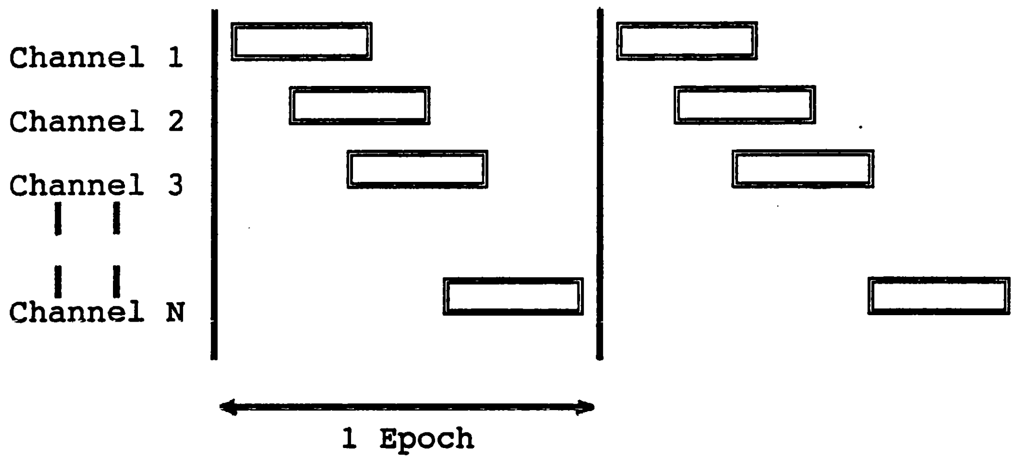


Figure 10.4. Timing of packet arrival.

Crystals in use at present, have a tolerance of ±0.1 ppm (see appendix VI). When used on a ring operating at 20 Mbps with 224 bit packets, the clock deviation at the end of each packet will be ±0.002% of a bit. This leads to a very low probability of losing any data.

However, the probability of packets shifting epochs is more significant and if the shift is in the wrong direction, data could be lost because of bunching. The length of of one epoch is given by;

$$1 \text{ epoch} = 24 / 12000 = 2 \text{ ms} \pm \delta\epsilon_1$$

where $\delta\epsilon_1$ is the deviation of an epoch, caused by clock slippage. A tolerance of ±0.1 ppm over a 2 ms period gives a possible error of ±0.2 ns. With each clock cycle taking 50 ns, a packet may fail to meet an epoch boundary every 250 epochs. Divided between 80 nodes, one operator may detect a slippage of one channel that he may be listening to, once every 40 seconds. This however, is the worst case. Half the errors will be in the opposite direction,causing bunching rather than spreading, which will have no ill effect in the long term. The only adverse effect of this action being to put a varying load on the DSP chip. In any case the mean frequency deviation in a crystal is zero.

The sampling clock being derived from a reconstituted 20MHz ring clock, means that samples occur periodically in synchronism with the network.

10.5 Voice Coding Timing Differences

Speech transfer quality has already been discussed in chapter 7. However, if the system were to be applied to a non-aircraft environment there would be a case for adopting an 8 kHz sampling rate rather than the 12 kHz proposed in this thesis. This would be compatible with ISDN (integrated services digital network) and other current telecom standards, using a 64 kbps serial bit rate instead of 96 kbps. Each packet would contain 3 ms of speech packet length time (T_{pl}) instead of 2 ms, therefore the maximum "open mic" capacity for a 20 Mbps ring will increase to 120 (multiply original number by 3/2). There is enough tolerance in the side-tone design limit to allow 3 ms plus ring access delay.

The delay for side tone corresponds to;

$$D_{st} \approx T_{pl} + (N_a \times 25.6) \mu s$$

where N_a is the number of active nodes on the ring. Also the equipment delay is negligible. Using this equation, it can be seen that while the proposed system meets the design criterion of 5 milliseconds, the ring can contain up to 117 nodes. However, if increased operator strain because of side-tone delay became acceptable, then D_{st} might be increased to 10 milliseconds [4] and an increase to 312 nodes can be achieved from a 20 Mbps ring.

The restrictions imposed by side-tone delay however, do not limit the design of the network unduly for the following reasons. The tolerable limit of 5 milliseconds imposed by side-tone delay can be increased to give a total ring delay of 100 milliseconds [4] if the side tone is generated locally. This new constraint on system delay (D_{st}) is no longer side-tone, but that of conversation delay. Therefore instead of 24 samples per packet, 1200 samples per packet may be used. The new ring bandwidth would be;

$$(9600 + 32) \times 80 / 0.1 = 7.7 \text{ Mbps}$$

instead of;

$$(192 + 32) \times 80 / 0.002 = 8.96 \text{ Mbps}$$

ie. for a 50 fold increase in point to point delay, only a reduction of 14% in bit rate is obtained.

10.6 Switching Time

The switching times for the contra-rotating ring to re-route itself will depend to a large extent on the equipment used to perform the operation and the processing device used to decide what action should be taken.

The time-out counter used to detect that no traffic is being received from the up-stream node waits for 25.6 µs (ie. 1 packet + 1 gap) before commencing self-healing. The total time for the loop-back operation to complete (T_{lb}) will therefore be;

$$T_{lb} = (2 \times 25.6) (N_c - 1) + T_d (N_c - 2) \ \mu s$$

where T_d is the time taken to make a loop-back decision and is software dependent, and N_c is the number of nodes connected to the ring. The (2 x 25.6) part of the equation is based on each node transmitting an "ok" packet then a "check" packet for all except the failed node, a 25.6 µs for the time-out counter and an intial 25.6 µs for the first "check" packet.

The ring effectively shuts down during the self-healing operation because no side-tone signal will pass through the faulty section of the ring.

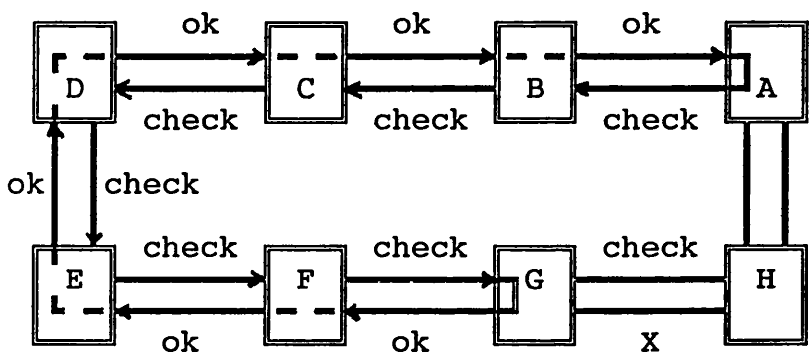


Figure 9.2. Loop-back activity.

From the data sheets in appendix VI, it can be seen that the total time for optical to electrical conversion and vice-versa will be less than 30 nanoseconds. If T_d is also ignored because it is significantly less than

25.6μs, the time taken for an 80 node ring to heal itself (T_{lb}) will be approximately 4 milliseconds. Figure 9.2 shows the configuration of a ring that has healed itself.

The time taken to select a bypass switch will be negligible in comparison to the loop back operation. However, some data will be lost as bypass selection is likely to occur the instant a node failure is detected.

10.7 Power-up

The method used for joining the ring at power-up is to monitor ring traffic, wait for a gap, then insert the register within 3.2 μs (the duration of a gap). The insertion of the register is performed accurately such that the optical input device has sufficient time to lock onto the packet pre-amble and synchronisation. This is achieved practically by counting the number of bits in the previous packet,

If a new node to the ring is the first to join, then the time-out operation will select monitor-mode (described in chapter 6) and that node will join the ring immediately without waiting for a gap. The node can detect that it is the first to join the ring by an absence of data on its inputs.

10.8 Summary

As expected, the Russell Ring performs adequately within the parameters that it was designed to meet. Also as expected, the performance of the ring degrades as its design parameters are exceeded.

11. RECOMENDATIONS FOR FUTURE WORK

An area of work that is continuing is to devise a node on the ring that can concurrently simulate the operation of several nodes. This will take the form of a communications services monitor (CSM) and provides the means for testing the ring to its full capacity and beyond without building in excess of 80 nodes. It will insert one packet every 2 ms for each node that is to be simulated, and its operation will be controlled by an IBM/PC. The CSM will be able to introduce known data errors into the ring and will therefore allow the system performance in the presence of errors to be assessed.

11.1 TEMPEST

TEMPEST considerations have not been explored fully in this thesis. Several options exist for implementing secure speech. Clear and secure speech could be routed using two separate contra-rotating rings with encrypted speech data routed using the clear ring. Alternatively, a single ring could be used for carrying both clear and secure speech with integrity checking to prevent security compromise. Thirdly, clear and secure speech could be carried on the same optical fibres using different light carrier frequencies.

11.2 Digital Filtering

The use of a digital signal processor (DSP) in each node facilitates the future development of digital filtering for automatic noise reduction/cancellation (ANR/C) and voice gain adjustment devices (VOGADs). This will reduce operator fatigue and improve speech intelligibility.

The DSP could also be used to detect quiet periods in order to reduce the bit rate. Some of the effects of bit rate reduction have already been discussed in chapter 10 and one way of achieving this is to not transmit during the quiet periods that are always present in conversations. This objective would not be easy to obtain, even if the reduction in network bandwidth was thought necessary, because a time reference would be required to indicate the duration of the quiet period to ensure correct decoding.

11.3 Speech Coding

Further reductions in network bandwidth may be obtained by the use of new speech coding schemes. The recent advent of ADPCM CODECs now makes it possible to implement in a small, easy-to-use package, and similar audio quality to that in the Demonstrator can be obtained in less than half the serial bit rate per channel.

11.4 Surface Mount Technology

To achieve compact nodes, surface mount technology could be used in a production system. This would give size and weight reduction. Alternatively, the use of hybrids may provide the required degree of compactness.

11.5 Large Packets

The transmission of large packets is not considered relevant to this thesis and so did not warrant investigation. However, should the need to transmit computer type data arise, it is an area recommended for future work. Combining slots to form large packets is thought to be the most suitable method, the only penalty being that the efficiency will be lower than for some other LANs.

11.6 DSQI

The Digital Speech Quality Index (DSQI) method for analysis and comparison of speech coding schemes is an area of study where a solution is urgently required. The lack of suitable comparison methods means that the evaluation of new coding schemes and implementaions always involves long and subjective testing. This is likely to be the most essential piece of future work resulting from this thesis.

12. CONCLUSIONS

The object of this thesis was to research and discuss a digital communication network for multi-seat aircraft. The network system was to be fault tolerant, simple to expand and modify, and should minimise non-recurring costs for each type of aircraft.

Established generations of aircraft CCS are largely centralised, with application specific central units, expansion difficulties and an upper limit on central unit capacity.

A requirement for simultaneous reception of multiple conversations together with other digital voice data considerations proved decisive when reasoning solutions to questions of TDM or FDM, control, topology and protocol.

The implementation of simultaneous reception of multiple conversations has been evaluated on digital signal processors (DSPs). Subjective tests have shown that the required speech quality can be maintained for a given maximum number of conversations. That maximum number depends on the DSP used.

The distributed network gives advantages of improved reliability, enhanced fault tolerance and better expandability. Digitized voice can give advantages over analogue voice, of reduced crosstalk, improved TEMPEST performance and provides the option of incorporating digital filtering and processing in the future.

Voice considerations eliminate the use of CCITT and IEEE 802 recommendations. An optical ring which incorporates the best features of the existing standards has been devised.

The use of optical fibres improves TEMPEST, EMC and because of recent developments, the radiation hardness required for military environments can be obtained. The long transmitter lifetimes required for a ring network are also now available.

Although a 4B-5B block code could have been used instead of Manchester II biphase, the 12.5 Mbps bit rate obtained with 4B-5B does not give any saving in terms of optical interface components compared with the 20 Mbps bit rate for Manchester II biphase. Therefore, the method that provides for the simplest clock recovery was chosen.

As can be seen from chapter 10 (performance), the capacity of the ring is the number of active nodes (open microphones). A 20 Mbps ring with 80 nodes connected can accommodate 78 open mics. Another limit is the address field (255 nodes) although for special applications, the source address field could run across into the special address field giving an absolute maximum of 2^{16} (65536) nodes. However, with each node contributing a delay of 25.6 μs, the total ring delay would be 1.6 seconds, therefore side-tone would need to be generated locally instead of being recovered after a full cycle of the ring. The last limit on ring size is the total ring delay. For the case where 5 ms total side-tone delay is the maximum, the number of nodes is given by:-

$$\frac{5 - 2}{0.0256} = 117 \text{ nodes}$$

where 2 ms is the delay in constructing a packet and 25.6 μs is the delay at each node. However, the effect of delay on side-tone is gradual and not absolute and therefore a ring larger than 117 nodes would gradually start to cause side-tone difficulties. This assumes that slots and packets are being used in the intended manner and not, for example cascading slots to form large packets.

Ring reliability has been increased by the use of a redundant ring. Simple uni-directional redundancy has the disadvantage that the failure of one node may make it impossible for others to transmit up-stream. Therefore, a contra-rotating ring was used, so that if one node fails totally, it can be removed from the ring by the nodes either side of it looping data flow back in the opposite direction by use of the second ring.

Although a 5 kHz audio bandwidth with 72 dB peak dynamic range has been adopted for the purpose of demonstration, which is thought to be more suitable for the high background noise environment of an aircraft, the system as it stands is able to accommodate a variation in the coding scheme adopted by trade-offs between parameters such as sampling-rate and side-tone delay. Despite attempts to adopt a single coding scheme this seems the best solution. Apart from the quality of reproduction associated with each coding scheme, the difference in operation of the ring would amount to different durations of speech being stored in each packet, which would always contain the same number of bits. For the case of digital telephony (64 kbps line rate per channel), 3 ms of speech would be stored in each packet.

The theories put forward in this thesis have been partially tested using the circuit boards shown in the photographs in appendix I. The prohibitive cost of the components in each node (approximately £1000) and the delivery time for the fibre-optic modules (which include Manchester coding and clock recovery), have made it impossible to test all the features of the system. However, the performance of features such as ring self-healing, ring synchronisation and control packets have been predicted in chapter 10. The design of the logic circuits within the gate-arrays has been proved, and packets can be assembled, passed to another node, then disassembled. Data transmission is shown in the scope pictures in appendix I.

The easy application of different coding techniques is made possible by the use of a layered approach to the system design. Although the 7-layer OSI reference model relates to computer systems, the rules can be adapted for other applications. The bottom 4 layers (transport) are used to map the network in a similar way to computer systems, but the top 3 layers (application) have been applied to the speech sub-system. This layered approach makes it possible to adapt large areas of design whilst keeping the existing system philosophy, and to incorporate bridges and/or gateways.

The system, as an implementation of the Russell Ring, has several advantages over conventional CCSs and over a CCS implemented with another LAN. These have been substantiated throughout this thesis. A key feature is the latency of the ring, which is bounded by design parameters that include an upper limit on the number of nodes that can transmit at a given instance. ie. if latency (side-tone) is flexible, then so is the number of nodes.

The installation of a Russell Ring into an aircraft, naval vessel or C^3 environment is one of the most important advantages as this will save users of the system weight, bulk and isolation, as well as the economic savings. One system has already been designed in anticipation for the requirements of a large military aircraft, a block diagram is shown in appendix VII.

13. ACKNOWLEDGEMENTS

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I must thank Mr J P Rendell (technical manager) for allowing me to publish company material and last but not least this thesis would not have been possible without the support of GEC Sensors Ltd.

14. REFERENCES

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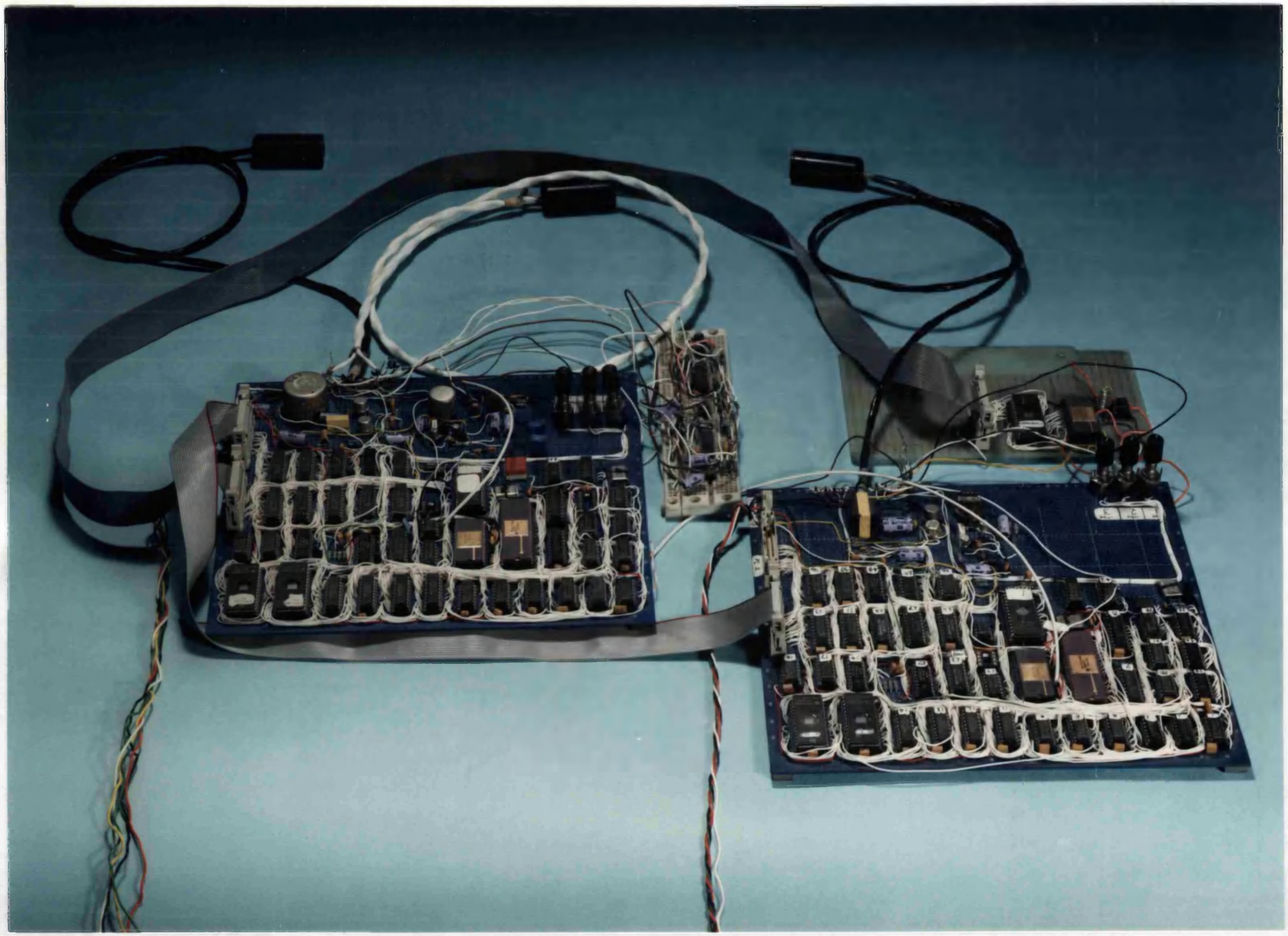
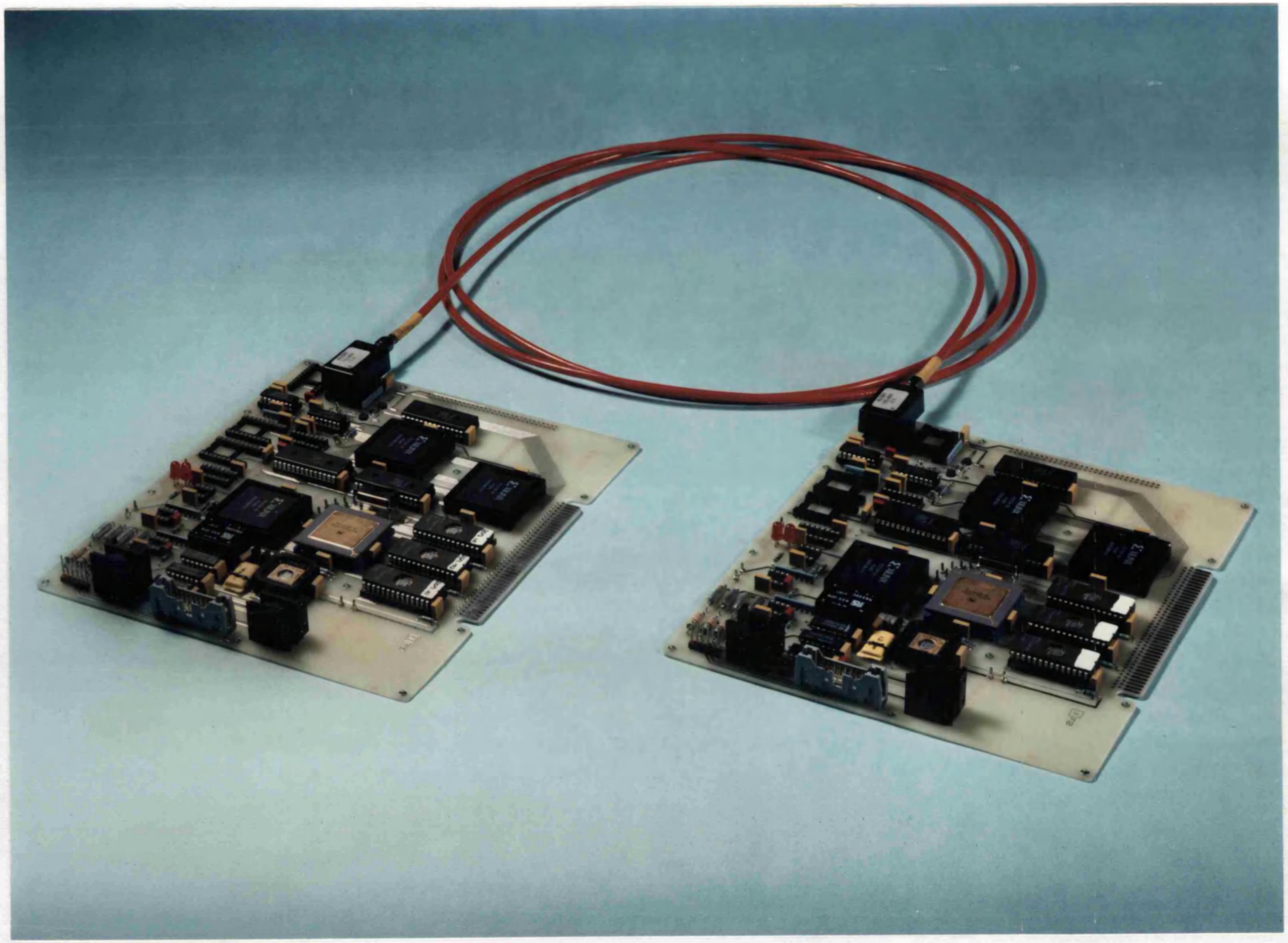
APPENDIX I

Photographs

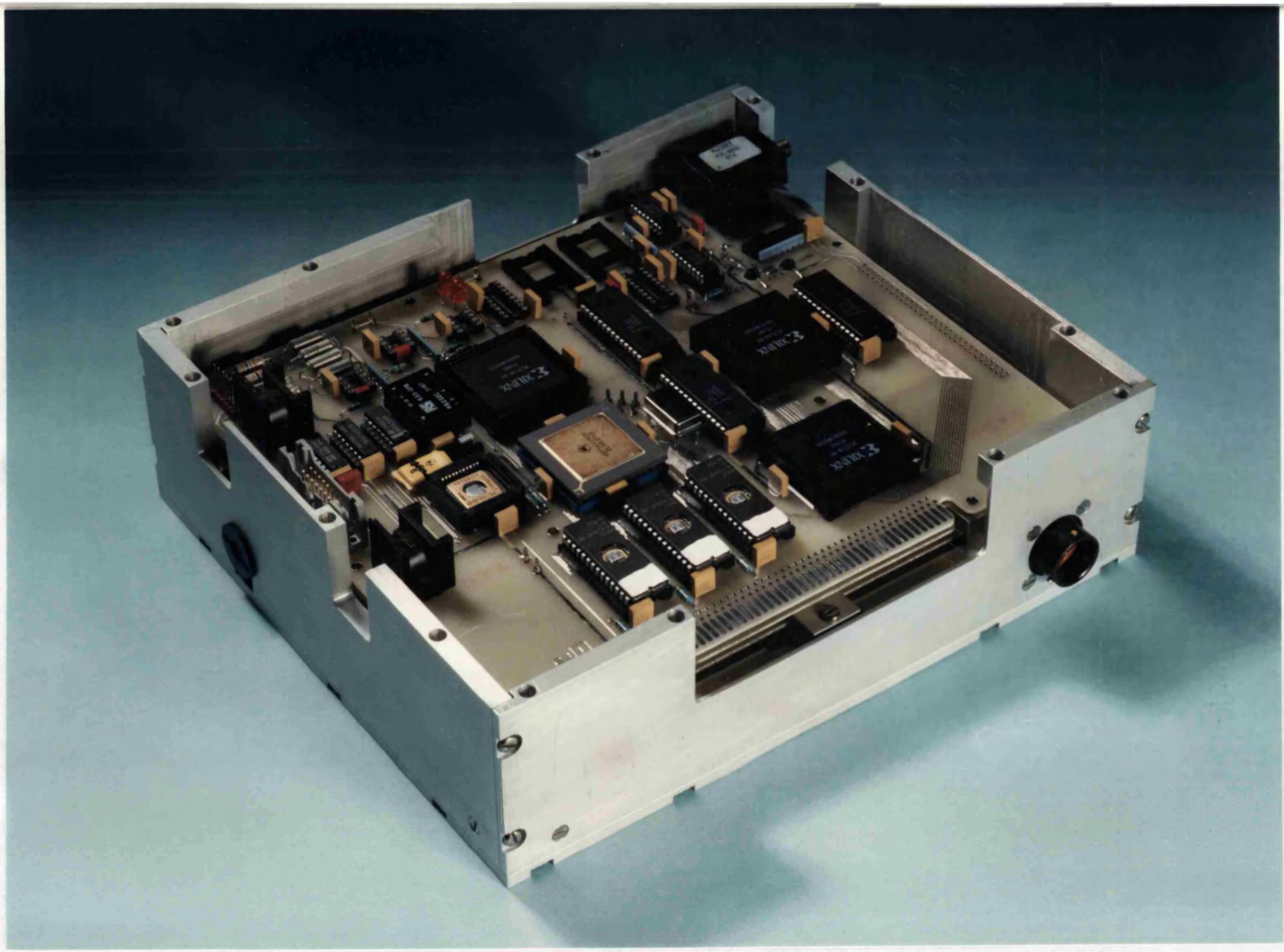


P1. SPE used for digitized speech evaluation.

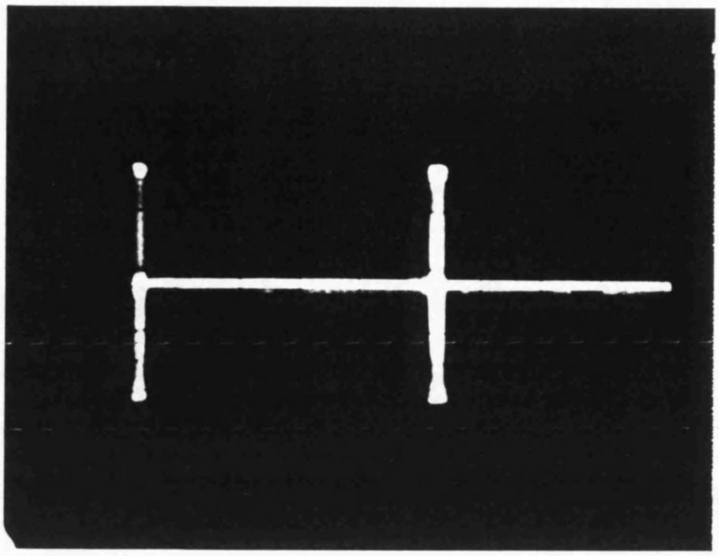


P2. Two core-cards connected via a fibre optic link.

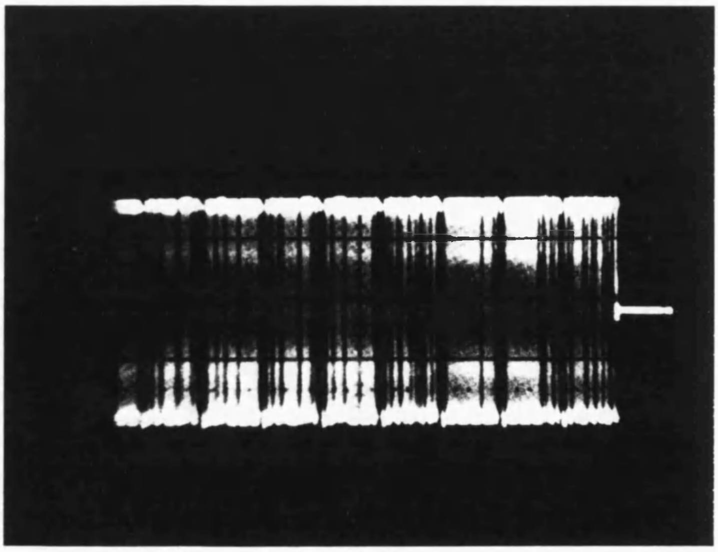
P3. Core-card mounted in development box with headset connected.



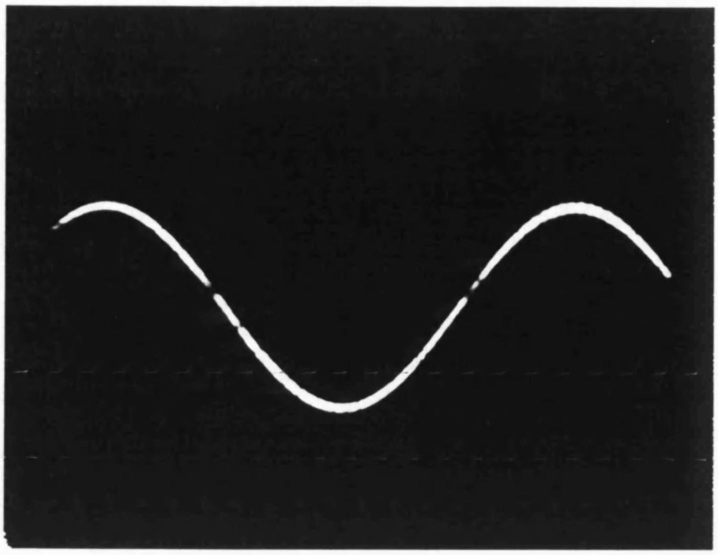
Close-up of core-card in development box.



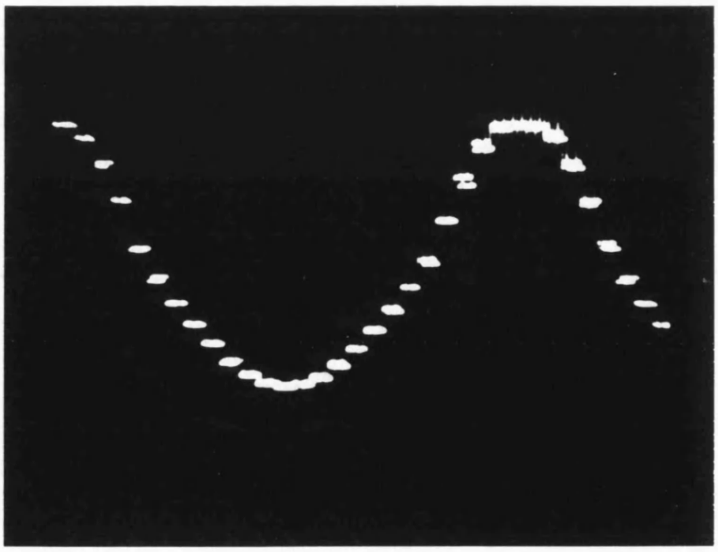
Scope trace showing the transmission of packets from one node, 2 ms apart.



One packet containing 224 bits.



Sine wave input at 500 Hz,

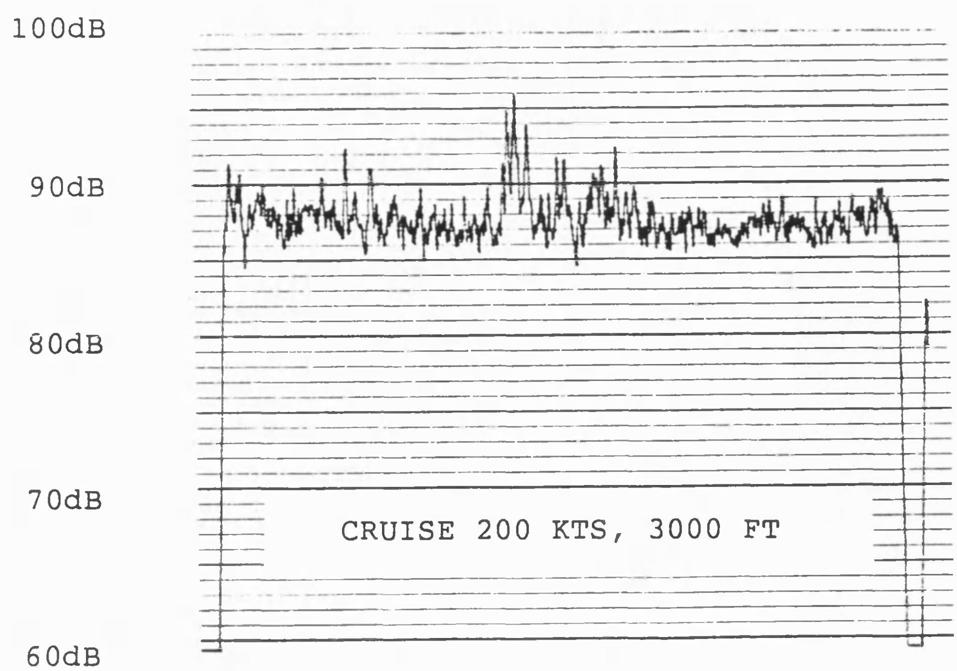


the same sine wave reconstructed at the output of a digital to analogue converter.

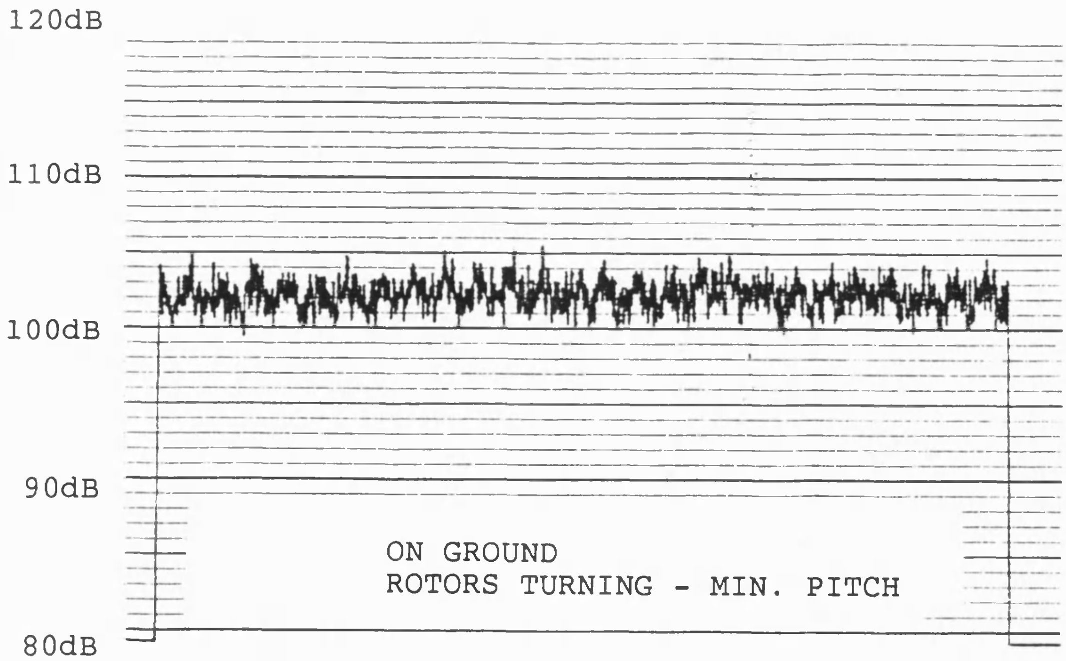
APPENDIX II

Noise Waveforms and

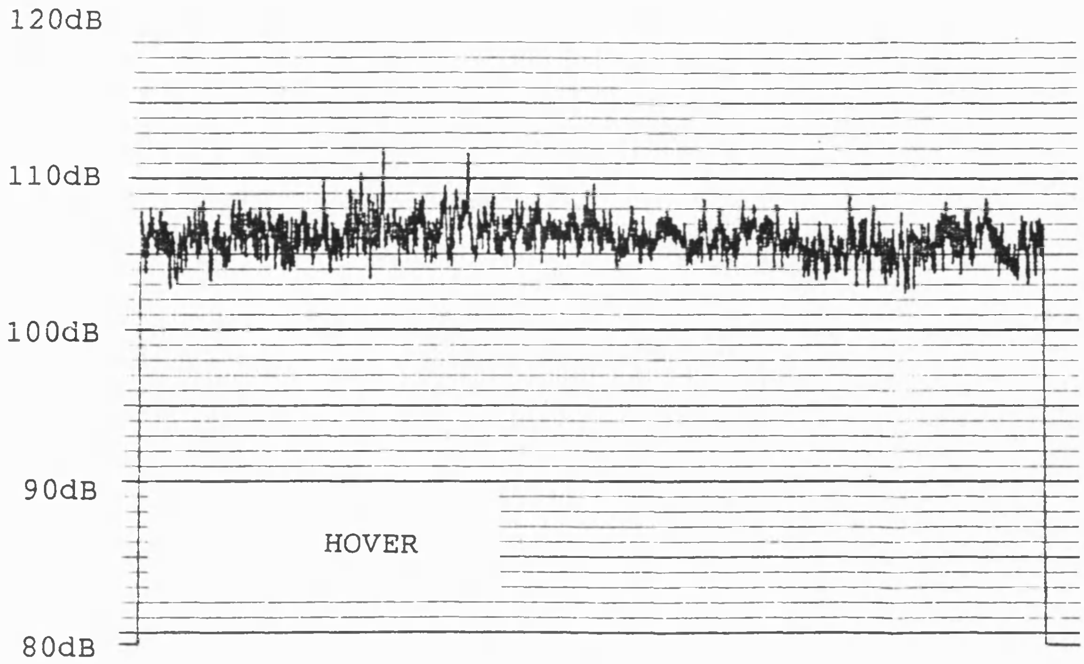
Test Sentences



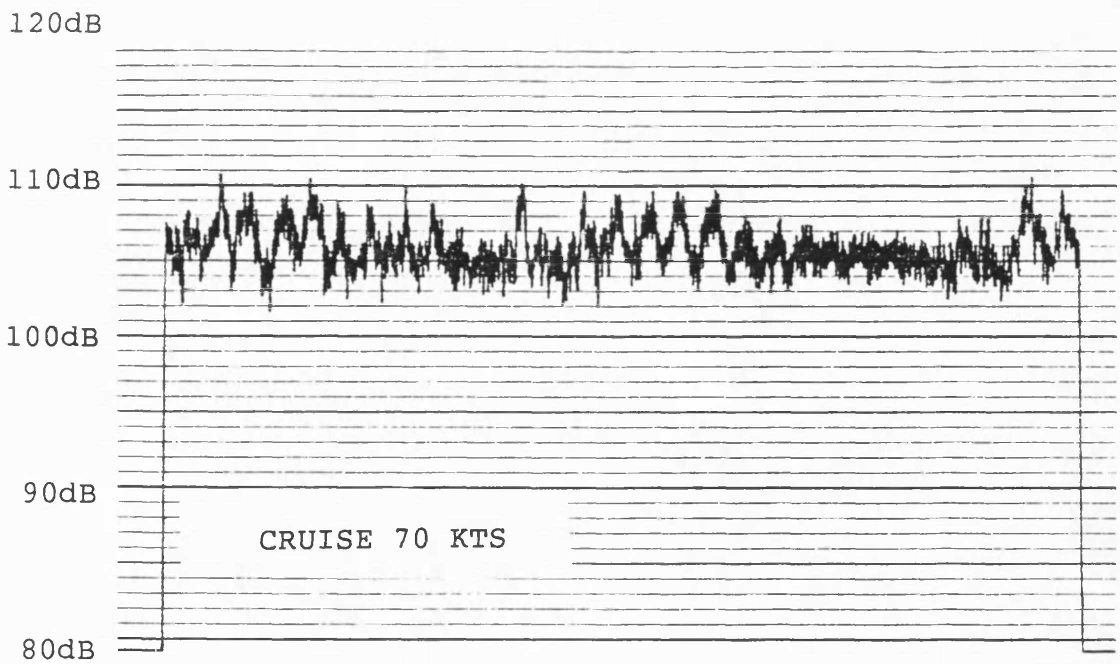
Cockpit acoustic noise for Maritime Reconnaisance Nimrod Mk I, flying at 200 knots and 3000 feet.



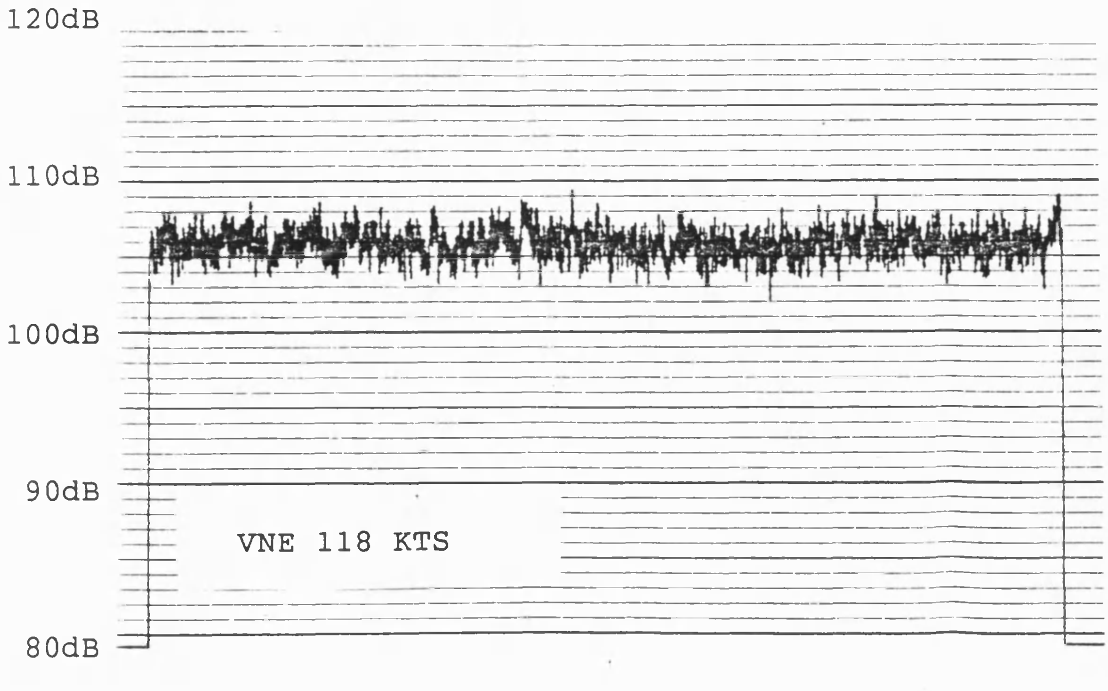
Cockpit acoustic noise for Sea King Mk 3.



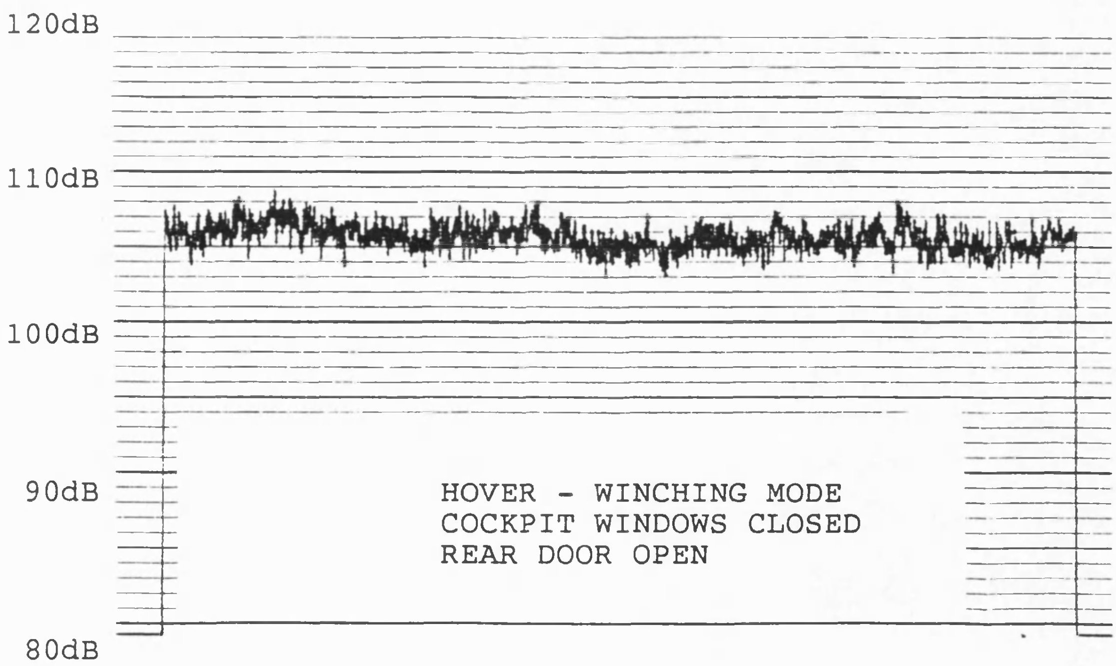
Cockpit acoustic noise for Sea King Mk 3.



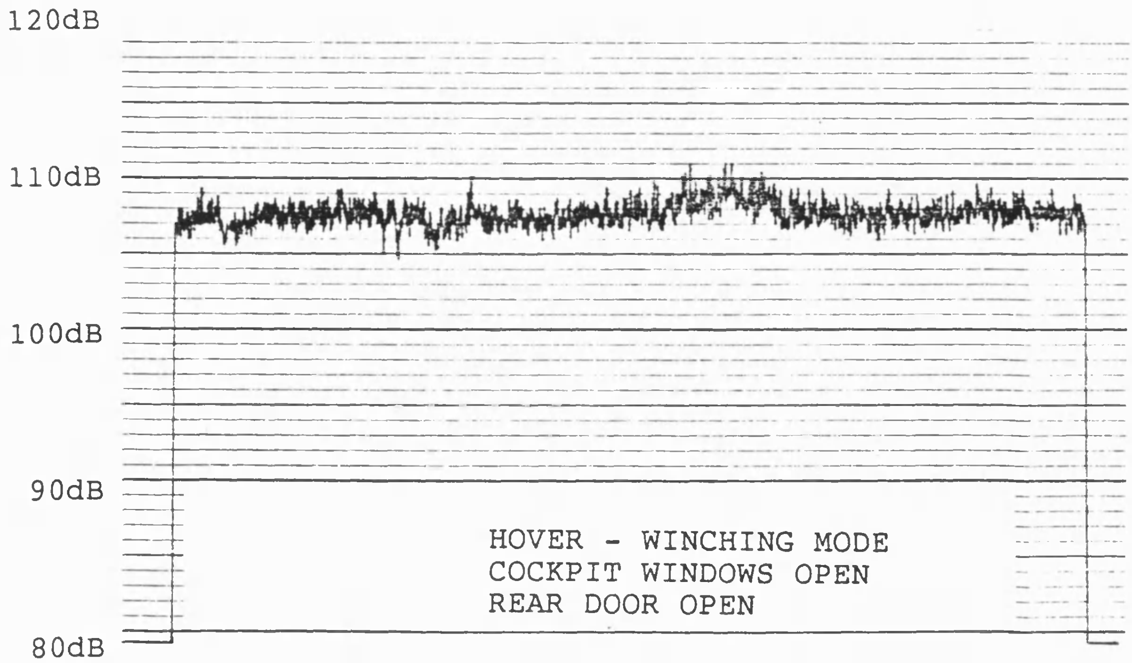
Cockpit acoustic noise for Sea King Mk 3.



Cockpit acoustic noise for Sea King Mk 3.



Cockpit acoustic noise for Sea King Mk 3.



Cockpit acoustic noise for Sea King Mk 3.

Message Number	Sentence
100	The chairs were of cracked, red leather
101	Its lights gleamed through a misty haze
102	The edge of his woollen coat was frayed
103	We can forage for food as we go
104	The engine warmed up and we were off
105	The teams played before a vast crowd
106	Fear is a great danger to peace
107	Plump for a job that needs courage
108	The luggage just sprawled in the aisles
109	Weeks went by without news of the men
110	There is zest in the work of jet flying
111	The forward cut in to score a goal
112	Holes were filled up with turf and heather
113	A staff car pulled up beside the truck
114	The small cramped room was cleanly kept
115	Treat broken skin for germs and dirt
116	The team started to train with a will
117	Address your ball with feet square
118	Joan came out in a blue and white affair
119	Use this and grease lifts off quickly
120	The new cast had trouble with the script
121	A flight of birds perched on the stone pillars
122	The wind rose swiftly to half a gale
123	My mount lashed out with gusto
124	Try a wooden one to reach the green
125	Tom pegged down the awning of the tent
126	I heard the soft sweet notes of the blackbird
127	Thick yarns are woven with fine ones
128	That golf game gives me the jitters
129	Wood pigeons nest in the sand dunes
130	Tom got in and twiddled all the knobs

Message Number	Sentence
131	The thief applied a drill to the lock of the safe
132	The lift will whisk you up in a second
133	Fan tailed doves strut about
134	Good manners are nice to have
135	The ball left the tee like a bullet
136	A white swan dived in the water weeds
137	The climbers spent a cold night on the ledge
138	The low soft warble of birds was clear
139	He fears the bills for imports will soar
140	The locals brought us eggs, milk and dates
141	Masts are seen above most roofs
142	Glazed fabric is a bright new vogue
143	A young man stood at the tiller of the barge
144	Remains of six more caves were found
145	The prince went on a world cruise with tutors
146	The sturdy colt won the mile in style
147	A chalk mark on the wall is the target
148	The cape coat was in proofed poplin
149	The book is tinged with grim humour
150	The stud suffers if the best are sold
151	The gardener broke the clod with a hoe
152	You cease to be a stranger once you come in
153	The fine view was both wild and lonely
154	The black mare broke into a trot
155	From a sharp ascent, we reached the edge
156	He stopped his digging to point the way
157	I planned a short cut by map and compass
158	Fill your can at the water tank
159	The clouds shed a purple light on the land
160	Their khaki shorts were black with mud
161	Blue birds pecked and strutted on the turf

Message Number	Sentence
162	The pools were seagreen shot with brown
163	The bells of flocks and heards sounded all round
164	Hundreds of red ants swarmed on the ground
165	The camp was in constant touch with the base
166	The bunks were hung with check curtains
167	Here was a friend at court to help matters
168	I wiped the new floor over with a cloth
169	Change took the matter in its hands
170	A strong east wind blew up the valley
171	A car with space to sleep is an idea
172	More patrols were called from their homes
173	A long metal tray was fitted at the table
174	The trees were hung with thick catkins
175	Tough clogs are good in wet weather
176	Wit and humour are the salt of our talk
177	A sense of perfect pitch is a rare gift
178	My front wheel hit the porch squarely
179	A gentle tug and the rope was hauled up
180	A tent is hard to pitch on stormy days
181	The cock first crows to announce the dawn
182	The sun had now left the highest peak
183	Bill chose a quiet spot for the camp
184	Plain cloths set off gay china
185	The spot was clearly marked for me
186	A warm rail near the bath is handy
187	He needs live tissue for his test
188	I took my bearings and drove off
189	White lines act as guides to drivers
190	He stayed on to face the crisis
191	Dick spotted the launch and struck out
192	Thick fog shrouded the line of huts
193	Narrow cups keep the tea warm

essage umber	Sentence
94	The white sheets are in soft nylon cloth
95	One sweet, one dry for this cocktail
96	The date palms were heavy with fruit
97	I gasped and plunged like a porpoise
98	Suits were of a fine corded weave
99	After each reel the film stopped
00	The skipper edged his craft to the bank
01	Apes with no tails are the missing link
202	She only now and then stops for breath
203	It is Ann's turn now to clear away
204	The pearl fishers dived for the last gem
205	The club had a swimming pool of sorts
206	It was no good trying to bluff him
207	Even the breeze burnt my arm
208	He clung hard to the running boards
209	Jack was small and dapper in a new suit
210	They stole three bottles of rum
211	The boy had a tousled mop of hair
212	They dropped bait in but never a bite
213	I closed by eyes to the blinding rays
214	The road had potholes two feet deep
215	Gay little foreign birds were seen
216	Cows, goats and dogs scatter for their lives
217	The mail car rattled down the rough road
218	Skip madly and in time you will slim
219	Read good verse to give rhythm
220	The role of the backs was in the main to defend
221	It was the best amoung a bunch of slick shows
222	Be your own skipper on the broads
223	My horse spilled me over its head
224	The stream wanders through bare hills
225	Ghosts are a relic of the dark ages

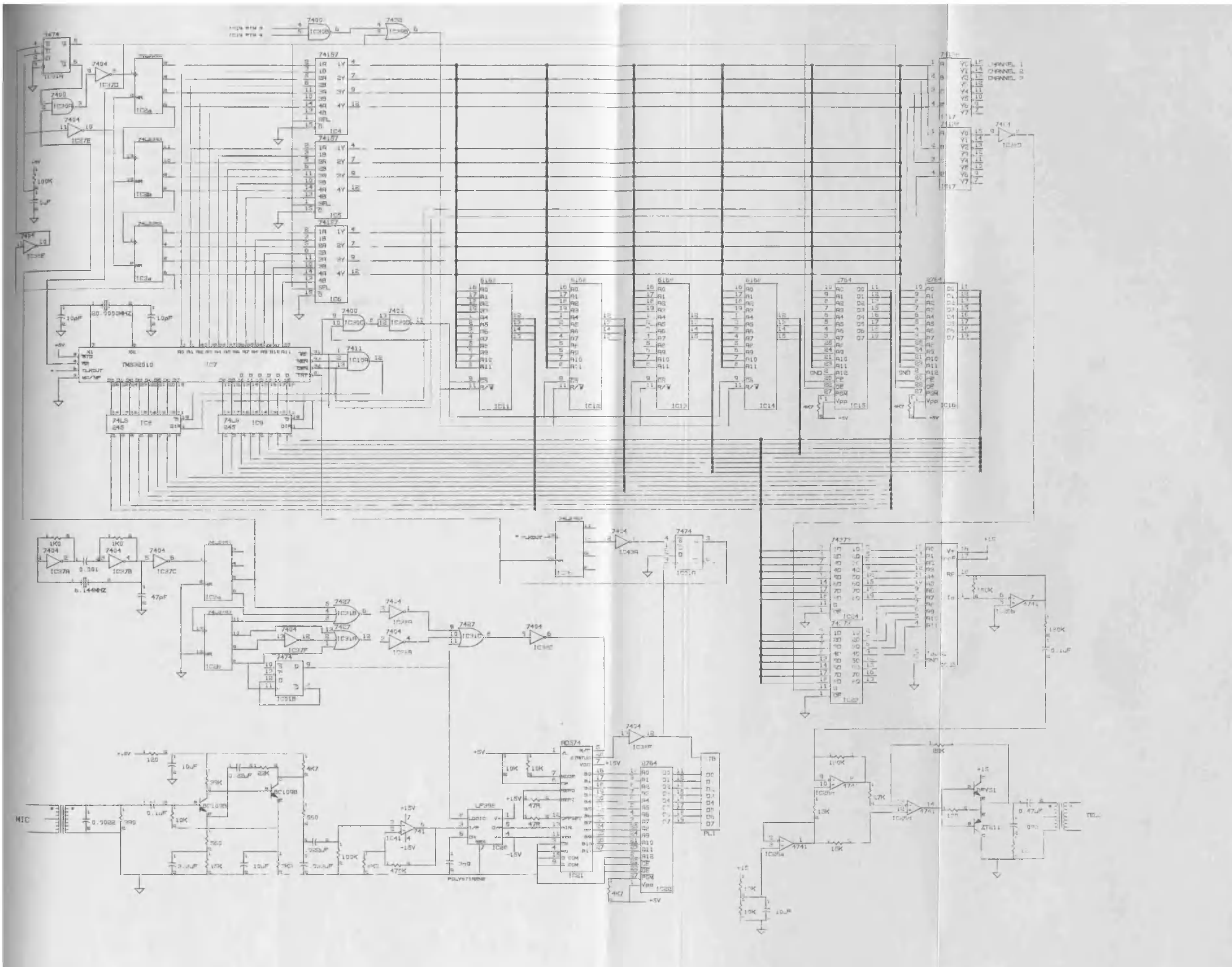
Message Number	Sentence
226	She wore a dress with a blue apron
227	Who slew the cat on a snowy day
228	Put your skates nicely in line
229	He wove a green cane basket
230	I recall with great glee my fight
231	The play has depth and charm and is well acted
232	The man shot the walrus from close range
233	The wool tunic was just hip length
234	It would suit me to go to warmer climets
235	My room mate was young, nice and cheerful
236	Groups of girls moped about in the rain
237	The trees and vines stretch across the road
238	Flags hung limp and droopy from the masts
239	The next lurch shot him into my arms
240	Out of clouds came broad shafts of sunlight
241	The first gong rang at seven bells
242	He puts a splash in things all-right
243	My horse shied at a roaring car
244	Camping out makes one fresh for the day
245	A dog walked quietly in and nipped the steak
246	His words fell like a blight on our spirits
247	The Arabs were tall with inky black beards
248	My fatigue and the sun put me to sleep
249	Through glasses, we saw huts ashore
250	I lost my trunk quite early on
251	Three dismal cheers were raised for his speech
252	The dog watched me eat with greedy looks
253	Hot food was served from field kitchens
254	The cargo was moved quickly to a key point
255	Dry and clear white wines are favoured
256	Long lines of poplars flank the route
257	I fished for trout in the small moorland streams

Message Number	Sentence
258	Storm clouds hung near the mountains
259	The stark record of the facts must be told
260	The chief dish was made with garlic and thyme
261	The rich grass raised good cattle
262	Ask for wine on draught, it is much cheaper
263	The white streaks are snow patches on the land
264	We have never lacked men of worth
265	He won a big prize for his new design
266	Tugs held the vessel as she neared the quay
267	Vacuum flasks keep your ice cool
268	Cars are driven up a ramp to the plane's door
269	Men feel fear in times of danger
270	The gang of men refused to load the van
271	He drew up quickly in his sports car
272	She hears with the deaf aid machine
273	The young man was armed with a rubber cosh
274	He just managed to land on the high cliffs
275	The farm wagon was drawn by a cart horse
276	Use the right coal and reduce smog
277	The air ferry will take the car for us
278	Wines are made from sunny big grapes
279	Art experts came to look at the fake
280	He got his degree by dint of hard work
281	Each case must be judged by its merits
282	His eyes reflect his clear vision
283	Riders and Hounds went at a slow pace
284	Grass tracks were made that covered miles
285	Make friends with all sorts of creatures
286	Tom motored south by a quick route
287	What's in the wind now at home and abroad
288	On one wall hung a coloured print
289	Three small lizards ran up the bark

Message Number	Sentence
290	Long stretches of sand lay to my left
291	Blue and gold fish were swimming in the bowl
292	His trousers were splashed with white paint
293	He crouched dripping by the bows of his yacht
294	At sea, we lived mostly on tinned foods
295	No words passed between us for some time
296	I have a car waiting in a back street
297	There is urgent need for long term plans
298	The police van screeched to a stop at the door
299	The same faces were grouped round the bar

APPENDIX III

SPE Circuit Diagram



CIRCUIT DIAGRAM FOR THE SPEECH PROCESSING EVALUATOR

APPENDIX IV

SPE Program

```
*________________________________________________
*
*    DCCS STATION PROGRAM
*    -=-=-=-=-=-=-=-=-=-=
*
*    WRITTEN BY     J.P.RUSSELL
*    DATE       13/12/85
*    VERSION    1.04
*
*________________________________________________
*
     IDT  'DIGITAL'
*
HDSET     EQU  0     *I/O output.
STAT1     EQU  0     *I/O inputs.
STAT2     EQU  1
STAT3     EQU  2
SAMP1     EQU  0     *memory locations.
SAMP2     EQU  1
SAMP3     EQU  2
VOL1 EQU  3
VOL2 EQU  4
VOL3 EQU  5
VOICE     EQU  6
HX7FF     EQU  7
FF   EQU  8
FFF  EQU  9
PRODL     EQU  10
PRODH     EQU  11
TEMP EQU  12
*
     AORG 0
*
     B    START
     B    INT
*
START     LDPK 0
*
     LACK >FF        *SAVE HEX FF,7FF & FFF FOR FUTURE USE.
     SACL FF
     SACL FFF
     LAC  FFF,4
     SACL FFF
     LACK >F
     ADD  FFF,0
     SACL FFF
     LAC  FFF,15
     SACH HX7FF,0
*

     EINT
LOOP B    LOOP
*
```

```
INT  NOP             *Interupt routine starts here. *
*RECEIVE AND STORE PARALLEL DATA
*AND VOLUME COEFFICIENTS.
*
     IN   SAMP1,STAT1    *start of program.
     IN   SAMP2,STAT2
     IN   SAMP3,STAT3
*
*====BODGE===============
*
     LAC  SAMP2,8
     SACH TEMP,0
     SACL SAMP2
     LAC  TEMP,0
     AND  FF
     ADD  SAMP2
     SACL SAMP2
*
     LAC  SAMP3,8
     SACH TEMP,0
     SACL SAMP3
     LAC  TEMP,0
     AND  FF
     ADD  SAMP3
     SACL SAMP3
*
*=========================
*
     LAC  SAMP1,6   *By loading the Acc. with
     SACH VOL1,0         *data and shifting it by 6,
     LACK >3F       *the top 16 bits will contain
     AND  VOL1      *the volume coefficient.
     SACL VOL1
     LACK >FF
     AND  SAMP1
     SACL SAMP1
*
     LAC  SAMP2,6
     SACH VOL2,0
     LACK >3F
     AND  VOL2
     SACL VOL2
     LACK >FF
     AND  SAMP2
     SACL SAMP2
*
     LAC  SAMP3,6
     SACH VOL3,0
     LACK >3F
     AND  VOL3
     SACL VOL3
     LACK >FF
     AND  SAMP3
     SACL SAMP3
```

```
*
*EXPAND A LAW TO LINEAR.
*
     LACK DECON
     ADD  SAMP1,0
     TBLR SAMP1
     LAC  SAMP1,0    *REMOVE DC OFFSET TEMPORARILY.
     SUB  HX7FF,0
     SACL SAMP1
*
     LACK DECON
     ADD  SAMP2,0
     TBLR SAMP2
     LAC  SAMP2,0
     SUB  HX7FF,0
     SACL SAMP2
*
     LACK DECON
     ADD  SAMP3,0
     TBLR SAMP3
     LAC  SAMP3,0
     SUB  HX7FF,0
     SACL SAMP3
*
*MULTIPLY SAMPLES WITH VOLUMES
*AND SUM THE THREE SAMPLES.
*
     LACK 0          *Set Acc. to zero.
*
     LT   SAMP1
     MPY  VOL1       *Multiply sample by volume then
     APAC            *add the product to the Acc.
*
     LT   SAMP2
     MPY  VOL2
     APAC
*
     LT   SAMP3
     MPY  VOL3
     APAC
*
     ADD  HX7FF,6    *THIS RE-INSERTS THE DC LEVEL.
*
     SACH PRODH,4    *Store Acc. temporarily.
     AND  FFF
     SACL PRODL
*
*SCALE AND ROUNDING.
*
     LAC  PRODL,10   *SCALING.
     SACH VOICE,0
     LAC  PRODH,6
```

```
     ADD  VOICE
     SACL VOICE
*
     BLZ  NEG        *ROUND OFF BOTTOM.
*
     LAC  VOICE,0
     AND  FFF        *ROUND OFF TOP.
     SUBS VOICE           *IS VOICE > FFF.
     BZ   OUTPUT
     LAC  FFF,0
*
     SACL VOICE           *Store Acc. as voice.
     B    OUTPUT
*
NEG  ZAC
     SACL VOICE
*
*OUTPUT TO D/A.
*
OUTPUT    OUT  VOICE,HDSET
*
     EINT            *End of interupt routine.
     RET

DECON     DATA  0
     DATA  86
     DATA  169
     DATA  248
     DATA  324
     DATA  397
     DATA  466
     DATA  533
     DATA  597
     DATA  658
     DATA  717
     DATA  773
     DATA  826
     DATA  878
     DATA  927
     DATA  975
     DATA  1020
     DATA  1063
     DATA  1105
     DATA  1144
     DATA  1182
     DATA  1219
     DATA  1254
     DATA  1287
     DATA  1319
     DATA  1350
     DATA  1379
     DATA  1408
     DATA  1435
     DATA  1460
```

```
DATA   1485
DATA   1509
DATA   1531
DATA   1553
DATA   1574
DATA   1594
DATA   1613
DATA   1631
DATA   1649
DATA   1666
DATA   1682
DATA   1697
DATA   1712
DATA   1726
DATA   1740
DATA   1753
DATA   1765
DATA   1777
DATA   1788
DATA   1799
DATA   1810
DATA   1820
DATA   1829
DATA   1838
DATA   1847
DATA   1856
DATA   1864
DATA   1871
DATA   1879
DATA   1886
DATA   1893
DATA   1899
DATA   1905
DATA   1911
DATA   1917
DATA   1923
DATA   1928
DATA   1933
DATA   1938
DATA   1942
DATA   1947
DATA   1951
DATA   1955
DATA   1959
DATA   1963
DATA   1966
DATA   1970
DATA   1973
DATA   1976
DATA   1979
DATA   1982
DATA   1985
DATA   1987
DATA   1990
```

DATA	1992
DATA	1994
DATA	1997
DATA	1999
DATA	2001
DATA	2003
DATA	2005
DATA	2006
DATA	2008
DATA	2010
DATA	2011
DATA	2013
DATA	2014
DATA	2016
DATA	2017
DATA	2018
DATA	2019
DATA	2021
DATA	2022
DATA	2023
DATA	2024
DATA	2025
DATA	2026
DATA	2027
DATA	2028
DATA	2029
DATA	2030
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DATA	2057

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DATA   2071
DATA   2072
DATA   2073
DATA   2075
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DATA   2078
DATA   2080
DATA   2081
DATA   2083
DATA   2084
DATA   2086
DATA   2088
DATA   2089
DATA   2091
DATA   2093
DATA   2095
DATA   2097
DATA   2100
DATA   2102
DATA   2104
DATA   2107
DATA   2109
DATA   2112
DATA   2115
DATA   2118
DATA   2121
DATA   2124
DATA   2128
DATA   2131
DATA   2135
DATA   2139
DATA   2143
DATA   2147
DATA   2152
DATA   2156
DATA   2161
DATA   2166
DATA   2171
DATA   2177
DATA   2183
```

```
DATA   2189
DATA   2195
DATA   2201
DATA   2208
DATA   2215
DATA   2223
DATA   2230
DATA   2238
DATA   2247
DATA   2256
DATA   2265
DATA   2274
DATA   2284
DATA   2295
DATA   2306
DATA   2317
DATA   2329
DATA   2341
DATA   2354
DATA   2368
DATA   2382
DATA   2397
DATA   2412
DATA   2428
DATA   2445
DATA   2463
DATA   2481
DATA   2500
DATA   2520
DATA   2541
DATA   2563
DATA   2585
DATA   2609
DATA   2634
DATA   2659
DATA   2686
DATA   2715
DATA   2744
DATA   2775
DATA   2807
DATA   2840
DATA   2875
DATA   2912
DATA   2950
DATA   2989
DATA   3031
DATA   3074
DATA   3119
DATA   3167
DATA   3216
DATA   3268
DATA   3321
DATA   3377
DATA   3436
```

```
DATA   3497
DATA   3561
DATA   3628
DATA   3697
DATA   3770
DATA   3846
DATA   3925
DATA   4008
DATA   4094
DATA   4095
```

APPENDIX V

BASIC Programs

Program for generating compression ROM

```
5LET V1MAX=2047
10LET V2MAX=127
15LET A=87.6 17LET LE=LOG(2.71828)
18DIM ROM(4095)
20PRINT"COMPANDING 12 BITS TO 8 BITS USING"
30PRINT"A COMPRESSION CHARACTERISTIC OF A=";A
40PRINT
45OPEN "CONV.ROM" FOR OUTPUT AS #1%
50PRINT
60FOR V1=0 TO 4095
70IF V1 <= V1MAX THEN GOTO 150
80X=(V1-V1MAX)/V1MAX
90IF X > 1/A THEN GOTO 120
100Y=(X*A)/(1+(LOG(A)/LE))
110GOTO 130
120Y=(1+(LOG(A*X)/LE))/(1+(LOG(A)/LE))
130V2=INT((Y*V2MAX)+V2MAX+0.5)
140GOTO 210
150X=(V1MAX-V1)/V1MAX
160IF X > 1/A THEN GOTO 190
170Y=(X*A)/(1+(LOG(A)/LE))
180GOTO 200
190Y=(1+(LOG(A*X)/LE))/(1+(LOG(A)/LE))
200V2=INT(V2MAX-(Y*V2MAX)+0.5)
210ROM(V1)=V2
220NEXT V1
310FOR INDEX = 0 TO 4095
320TEMP1 = (ROM(INDEX)+1)/16
330IF TEMP1 < 1 THEN 800
340TEMP2 = INT(TEMP1)
350TEMP4 = TEMP2
360GOSUB 1000
370HEXNUMBER$ = HEXDIGIT$
380TEMP4 = (ROM(INDEX)+1)-(TEMP2*16)
390GOSUB 1000
400HEXNUMBER$ = HEXNUMBER$+HEXDIGIT$
790GOTO 820
800HEXNUMBER$="0"
803TEMP4 = ROM(INDEX)+1
805GOSUB 1000
810HEXNUMBER$ = HEXNUMBER$+HEXDIGIT$
820PRINT #1%,HEXNUMBER$
870NEXT INDEX
990CLOSE CONV.ROM
999GOTO 10000
```

```
1000IF TEMP4 > 15 THEN PRINT "ERROR, TEMP4 = ";TEMP4
1010IF TEMP4 < 10 THEN HEXDIGIT$ = STR$(TEMP4)
1020IF TEMP4 = 10 THEN HEXDIGIT$ = "A"
1030IF TEMP4 = 11 THEN HEXDIGIT$ = "B"
1040IF TEMP4 = 12 THEN HEXDIGIT$ = "C"
1050IF TEMP4 = 13 THEN HEXDIGIT$ = "D"
1060IF TEMP4 = 14 THEN HEXDIGIT$ = "E"
1070IF TEMP4 = 15 THEN HEXDIGIT$ = "F"
1080RETURN
10000END
```

Program for generating 'A' law expansion table

```
5  LET A=87.6
10 LET V1HALF=2047
15 LET V2HALF=127
17 LET LE=LOG(2.71828)
20 PRINT "DE-COMPANDING 8 BITS TO 12 FOR USE IN THE TMS320,"
25 PRINT "WITH A COMPRESSION CHARACTERISTIC OF A=";A
30 PRINT
40 OPEN "DECON.ROM" FOR OUTPUT AS #1%
50 PRINT
60 FOR V2=0 TO 255
70IF V2 <= V2HALF THEN GOTO 150
80Y=(V2-V2HALF)/V2HALF
90IF Y > 16/A THEN GOTO 120
100X=(Y/A)*(1+(LOG(A)/LE))
110GOTO 130
120X=(2.71828**(Y*(1+(LOG(A)/LE))-1))/A
130V1=INT((X*V1HALF)+V1HALF+0.5)
140GOTO 210
150Y=(V2HALF-V2)/V2HALF
160IF Y > 16/A THEN GOTO 190
170X=(Y/A)*(1+(LOG(A)/LE))
180GOTO 200
190X=(2.71828**(Y*(1+(LOG(A)/LE))-1))/A
200V1=INT(V1HALF-(X*V1HALF)+0.5)
210PRINT #1%,"DATA ";V1
220 NEXT V2
230 CLOSE DECON.ROM
1000 END
```

Program to generate TMS320 format files

```
1PRINT
2PRINT"FORMATTER"
3PRINT
4PRINT"Input file is in hex format and output file"
5PRINT"is in TMS9900 format."
6PRINT"The output can be converted to TEKHEX by"
7PRINT"using RUN OBJCNV only if used on the VAX."
8PRINT
10OPEN "CONV.ROM" FOR INPUT AS #2%
15OPEN "CONVERT.TI" FOR OUTPUT AS #3%
17FFFF% = 65535
18F% = 15

20REM CONSTRUCT WHOLE HEX FILE ON 1 LINE
25LINE$ = "00000CONVERT 90000"
30FOR INDEX1 = 0 TO 4095 STEP 2
40LINE$ = LINE$ + "B"
50FOR INDEX2 = 1 TO 2
60INPUT #2%, TEMP$
70LINE$ = LINE$+TEMP$
80NEXT INDEX2
90NEXT INDEX1

100REM 1st LINE CHECKSUM
110TMPLN$ = LEFT$(LINE$,63)+"7"
120NEWLN$ = TMPLN$
130CHKSM% = 0
140FOR INDEX3% = 1 TO 64
150CHKSM% = CHKSM%+ASCII(TMPLN$)
160TMPLN$ = RIGHT$(TMPLN$,2%)
170NEXT INDEX3%
180GOSUB 1000

300REM CHECKSUM FOR MAIN HEX FILE.
310REM REMEMBERING THAT THE 1st LINE WAS
320REM TWO CHARACTERS SHORTER.
330MAIN$ = "XX"+LINE$
350FOR INDEX5% = 1 TO 157
400MAIN$ = RIGHT$(MAIN$,66)
410TMPLN$ = LEFT$(MAIN$,65)+"7"
420NEWLN$ = TMPLN$
430CHKSM% = 0
440FOR INDEX4% = 1 TO 66
450CHKSM% = CHKSM%+ASCII(TMPLN$)
460TMPLN$ = RIGHT$(TMPLN$,2%)
470NEXT INDEX4%
480GOSUB 1000
490NEXT INDEX5%
500PRINT #3%,":COMPANDER (C) JPR SEPT 85"
```

```
990GOTO 10000

1000REM HEX CHECKSUM WITH 2's COMPLIMENT GENERATOR
1010HXCHSM$ = ""
1020CHKSM% = (CHKSM%-1%) XOR FFFF%
1030FOR I% = 1 TO 4
1040DIGIT = (CHKSM%/(16%**(4%-I%))) AND F%
1050GOSUB 1100
1060HXCHSM$ = HXCHSM$+HEXDIGIT$
1070NEXT I%
1080NEWLN$ = NEWLN$+HXCHSM$+"F"
1090PRINT #3%,NEWLN$
1095RETURN

1100IF DIGIT < 10 THEN HEXDIGIT$ = STR$(DIGIT)
1110IF DIGIT = 10 THEN HEXDIGIT$ = "A"
1120IF DIGIT = 11 THEN HEXDIGIT$ = "B"
1130IF DIGIT = 12 THEN HEXDIGIT$ = "C"
1140IF DIGIT = 13 THEN HEXDIGIT$ = "D"
1150IF DIGIT = 14 THEN HEXDIGIT$ = "E"
1160IF DIGIT = 15 THEN HEXDIGIT$ = "F"
1170RETURN

10000CLOSE CONVERT.TI
10010CLOSE CONV.ROM
10020END
```

```
10 CLS
20 PRINT "*************************************************"
30 PRINT "* TMS320 TO DATA IO CONVERTER    10/7/86        *"
50 PRINT "*                                               *"
60 PRINT "*                                               *"
70 PRINT "* WRITTEN BY J.P.RUSSELL BSc,MSc,CEng,MIEE      *"
80 PRINT "*                                               *"
85 PRINT "* THE DATA IO MUST BE SET UP IN BINARY FORM AND *"
87 PRINT "* OF THE DEVICE TYPE OF YOUR CHOICE........THIS *"
90 PRINT "* PROGRAM GENERATES A .HEX FILE FOR READING, AND*"
92 PRINT "* .DHI AND .DLO FILES FOR PROGRAMING.           *"
95 PRINT "*                                               *"
100 PRINT "*************************************************"
120 PRINT "WHICH FILE WOULD YOU LIKE ME TO CONVERT";
140 SOUND 523,5
150 INPUT P$
160 N=0
170 N=N+1
180 A$=MID$(P$,N,1)
190 IF A$<>"." THEN B$=LEFT$(P$,N):GOTO 170
199 R$="            "
200 MID$(R$,1)=B$
210 MID$(R$,N)=".HEX"
220 PRINT R$" CREATED"
221 S$="            "
222 HI$="            "
223 LO$="            "
224 MID$(S$,1)=B$
225 MID$(HI$,1)=B$
226 MID$(LO$,1)=B$
227 MID$(S$,N)=".DIO"
228 MID$(HI$,N)=".DHI"
229 MID$(LO$,N)=".DLO"
230 PRINT S$" CREATED"
231 PRINT HI$" CREATED"
232 PRINT LO$" CREATED"
260 SOUND 587,10
310 PRINT"CONVERSION STARTED"
320 OPEN "I",#1,P$
330 OPEN "O",#2,R$
340 OPEN "O",#3,S$
350 ADDR = 0
360 LINE INPUT #1,FRED$
370 IF LEFT$(FRED$,1) = "9" THEN 820
390 IF LEFT$(FRED$,1) = "B" THEN 450
400 IF LEFT$(FRED$,1) = "7" THEN 790
410 IF ASC(LEFT$(FRED$,1)) = 10 THEN 790
420 IF LEFT$(FRED$,1) = ":" THEN 670
430 FRED$ = MID$(FRED$,2)
440 GOTO 370
450 PRINT #2,MID$(FRED$,2,4);
```

```
460 A = ASC(MID$(FRED$,2,1))
470 B = ASC(MID$(FRED$,3,1))
480 C = ASC(MID$(FRED$,4,1))
490 D = ASC(MID$(FRED$,5,1))
500 A = A - 48 510 B = B - 48
520 C = C - 48 530 D = D - 48
540 IF A > 9 THEN A = A - 7
550 IF B > 9 THEN B = B - 7
560 IF C > 9 THEN C = C - 7
570 IF D > 9 THEN D = D - 7
580 E = B + 16 * A
590 F = D + 16 * C
600 E$ = CHR$(E)
610 F$ = CHR$(F)
620 PRINT #3,E$;F$;
640 FRED$ = MID$(FRED$,6,LEN(FRED$))
650 ADDR = ADDR + 1
660 GOTO 370
670 PRINT #3,"JOHN"
690 PRINT "SPLITTING STARTED"
700 SOUND 659,10
775 CLOSE #1,#2,#3
780 REM   NOW JUMP TO SPLITTING PROGRAM
785 GOTO 10000
790 PRINT #2,
810 GOTO 360
820 W = ASC(MID$(FRED$,2,1))
830 X = ASC(MID$(FRED$,3,1))
840 Y = ASC(MID$(FRED$,4,1))
850 Z = ASC(MID$(FRED$,5,1))
860 W = W - 48 870 X = X - 48
880 Y = Y - 48 890 Z = Z - 48
900 IF W > 9 THEN W = W - 7
910 IF X > 9 THEN X = X - 7
920 IF Y > 9 THEN Y = Y - 7
930 IF Z > 9 THEN Z = Z - 7
940 AORG = Z + (16 * Y) + (16 * 16 * X) + (16 * 16 * 16 * W)
950 LOOPNO = AORG - ADDR
960 IF LOOPNO = 1 OR LOOPNO < 1 THEN 1030
970 FOR S = 1 TO LOOPNO 9
80 PRINT #3,CHR$(127);CHR$(128);
990 PRINT #2,CHR$(55);CHR$(70);CHR$(56);CHR$(48);
1000 PRINT "7F80";
1010 ADDR = ADDR + 1
1020 NEXT S
1030 FRED$ = MID$(FRED$,6,LEN(FRED$))
1040 GOTO 370
10000 OPEN "I",#1,S$
10010 OPEN "O",#2,HI$
10020 OPEN "O",#3,LO$
10022 MARKER = 0
10023 BYTE$ = INPUT$(3,#1)
10025 THING$=INPUT$(1,#1)
```

```
10030 BYTE$ = BYTE$ + THING$
10050 IF BYTE$ = "JOHN" THEN 11000
10052 IF MARKER = 1 THEN 10082
10059 PRINT #2, LEFT$(BYTE$,1);
10060 MARKER = 1
10070 BYTE$ = RIGHT$(BYTE$,(LEN(BYTE$)-1))
10080 GOTO 10025
10082 PRINT #3, LEFT$(BYTE$,1);
10089 MARKER = 0
10090 BYTE$ = RIGHT$(BYTE$,(LEN(BYTE$)-1))
10100 GOTO 10025
11000 CLOSE #1
11010 CLOSE #2
11020 CLOSE #3
11030 SOUND 523,2
11040 SOUND 587,2
11050 SOUND 659,2
11060 SOUND 698,2
11070 SOUND 783,2
11080 SOUND 880,2
11090 SOUND 987,2
11100 SOUND 1046,2
11110 PRINT"CONVERSION COMPLETED"
11120 END
```

APPENDIX VI

Data Sheets

DATA
FROM
CELDIS
(0734) 586191

UNITED TECHNOLOGIES MOSTEK

COMMUNICATION PRODUCTS

PRELIMINARY DATA SHEET

A-LAW COMPANDING CODEC WITH FILTERS MK5356(P/N/J)/MK5326(P/N/J)

FEATURES

- Complete per-channel, single chip CODEC with filters
- Receive power amplifiers (MK5326)
- D3/D4 and CCITT compatible
- On-chip voltage references
- ±5 volt power supplies, ±5%
- Low power dissipation
  - 40mW typical without power amplifiers
  - 1 μW typical in power-down mode
- Innovative clock design well-suited for digital telephone applications
- TTL/CMOS-compatible digital inputs and outputs
- Gain adjust available at the transmit and receive filter stages
- Synchronous or asynchronous operation
- Serial data rate from 64 kb/s to 4.096 Mb/s
- Separate internal analog and digital grounds reduce system noise problems

DESCRIPTION

The MK5356 and MK5326 are silicon-gate CMOS devices containing a companding CODEC and PCM filters on a single chip. The MK5326 also contains receive power amplifiers. These devices meet the needs of the telecommunications industry for per-channel, voice-frequency CODECs and PCM filters. The complete transmit and receive functions have been incorporated into a single package with a high degree of crosstalk immunity. Typical applications are channel banks, central offices, digital telephones, PBX systems, ISDN and other telephone digital switching and transmission systems.

An input amplifier, band-pass filter and compressing A/D converter make up the MK5356/MK5326 transmit

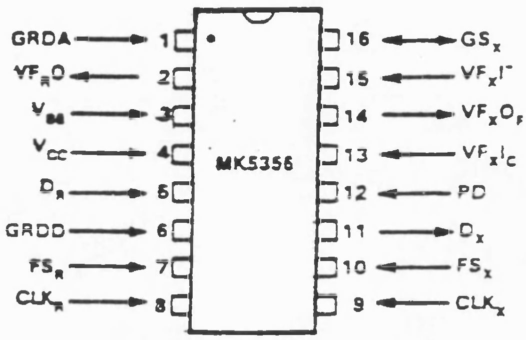


Figure 1. Pin Connections

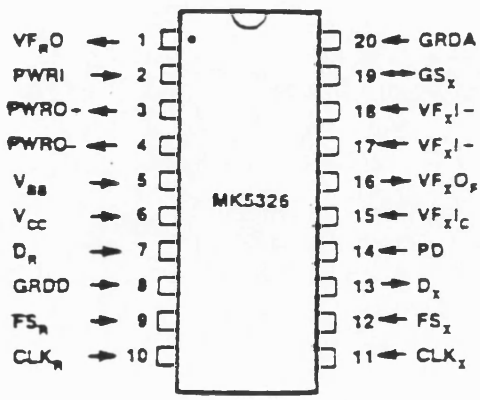


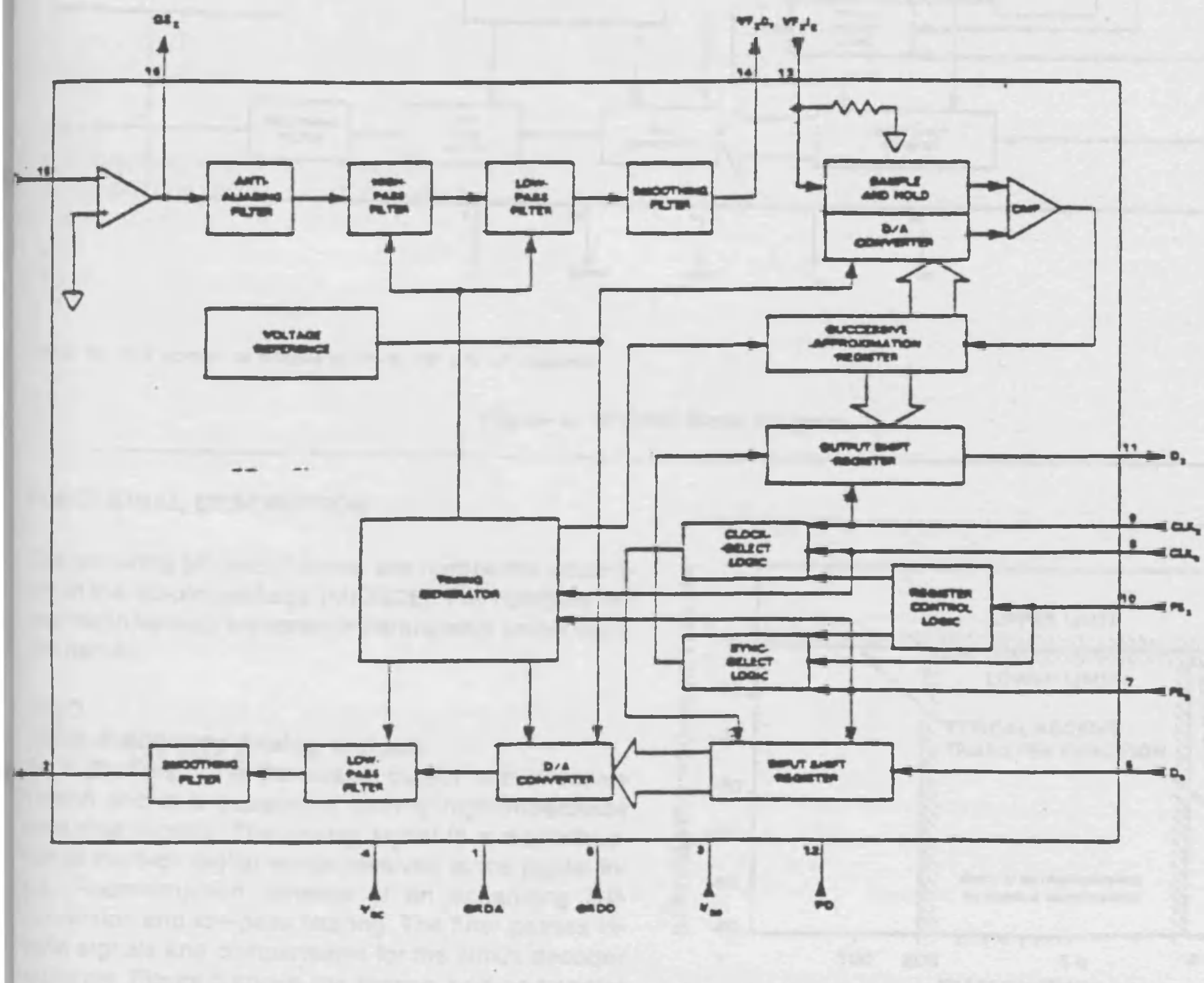
Figure 2. Pin Connections

tions. The operational amplifier output is available use in a gain configuration at the transmit stage, in ition to the 6 dB gain in the transmit filter. By bling the amplifier, this output serves as a noninvert- high-impedance input to the band-pass filter. The d-pass, switched-capacitor filter provides rejection he 50-60 Hz power line frequency and the band- ting required for an 8 kHz sampling system. The A/D verter transforms the band-limited, voice-frequency nals into 8-bit words. A data clock and frame syn- nization input control the serial transmission of en- ed data.

MK5356 receive section is made up of an expand- D/A converter and a low-pass filter. The MK5326 ive section contains the same components plus a differential power amplifer pair.

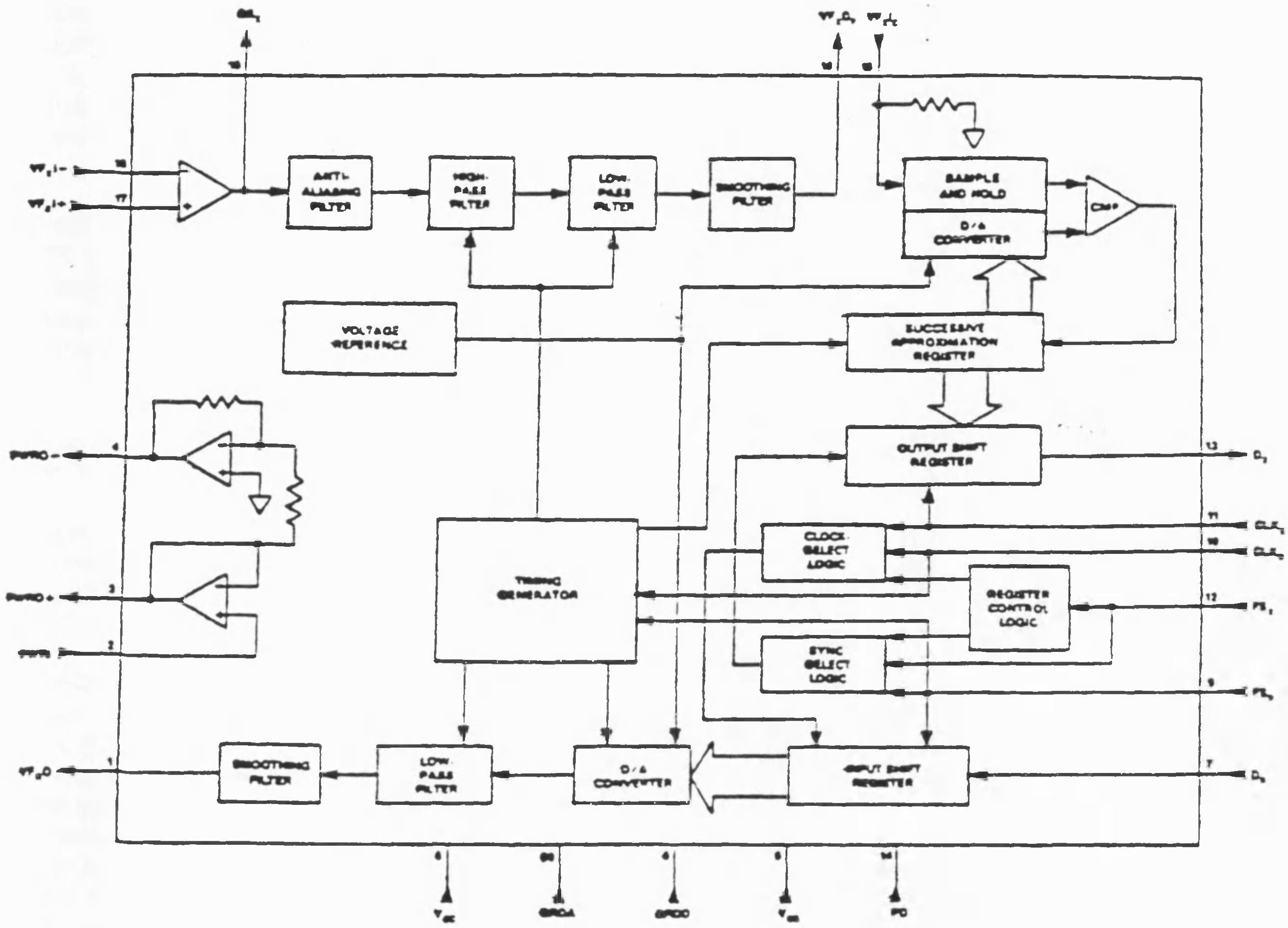
The D/A converter receives 8-bit words in a serial format controlled by the data clock and a frame synchronization input. The low-pass, switched-capacitor filter smooths the D/A converter voltage steps and provides compensation for the sinx/x decoder response. A voltage divider network may then be used to adjust the receive filter output to system levels. On the MK5326, the differential power amplifier pair is available for low-impedance drive capability applications.

Pin connections for the MK5356/MK5326 are shown in Figures 1 and 2. A block diagram of the MK5356 is shown in Figure 3. Figure 4 shows the block diagram of the MK5326.



Pin 13 is typically ac coupled to Pin 14 with a 0.1 µF capacitor.

Figure 3. MK5356 Block Diagram



NOTE: Pin 15 is typically ac coupled to Pin 16 with a 0.1 μF capacitor.

Figure 4. MK5326 Block Diagram

FUNCTIONAL DESCRIPTION

The following pin descriptions are numbered according to the 20-pin package (MK5326). Pin numbers for the 16-pin version are listed in parenthesis under each pin name.

VF_RO
(Voice-Frequency Analog Output)

Pin 1 (2). This pin is the analog output of the receive section and it is capable of driving high-impedance electronic hybrids. The analog signal is a reconstruction of the 8-bit digital words received at the digital input. Reconstruction consists of an expanding D/A conversion and low-pass filtering. The filter passes in-band signals and compensates for the sinx/x decoder response. Figure 5 shows the receive section transfer characteristics.

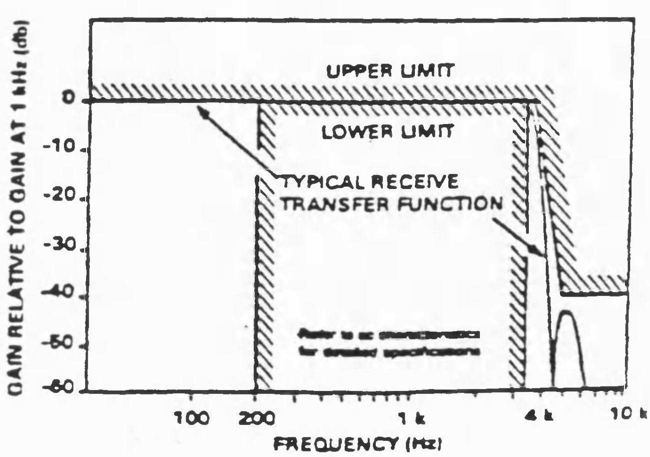


Figure 5. Receive Transfer Characteristics

The load connected to VF_RO should not exceed 10 kΩ in parallel with 30 pF. The output voltage range is ±3.2 V. In MK5326 applications requiring low-impedance drive capability, the differential power amplifier pair may be utilized by connecting VF_RO to PWRI. The receive signal is then taken from the power amplifier outputs PWRO+ and PWRO−.

PWRI
(Power Amplifier Input)
Pin 2. PWRI is the input to the receive section's power amplifier stage. A resistor divider may be used on VF_RO to adjust the level at PWRI. A differential pair of output signals is then available at PWRO+ and PWRO−.

Alternate Function — The power amplifier stage can be deactivated by connecting PWRI to V_{BB}. In this mode, the amplifiers are disabled and device power consumption is reduced. If PWRI is connected to V_{BB} with power supplies connected, the MK5326 should be cycled through the Power-Down or Standby Mode to insure reactivation of the power amplifier stage.

PWRO+ AND PWRO−
(Power Amplifier Outputs)
Pins 3 and 4. Pins 3 and 4 provide a balanced differential output from the power amplifier stage of the receive section. This section is provided for applications requiring low impedance drive capability. PWRO+ and PWRO− can supply a differential signal of up to ±6.4 V into 600Ω or a single-ended signal swing of up to ±3.2 V into 300Ω loads referenced to analog ground.

V_{BB}
(Negative Power Supply Input)
Pin 5 (3). This pin should be supplied with a voltage of −5 V ± 5% with GRDA and GRDD providing the ground return connections.

V_{CC}
(Positive Power Supply Input)
Pin 6 (4). This pin should be supplied with a voltage of +5 V ± 5% with GRDA and GRDD providing the ground return connections.

(Data Serial Input)
Pin 7 (5). This pin provides the digital input to the receive data register. The input register accepts an 8-bit data word and loads it under control of a data clock and the receive frame synchronization input. When FS_X is connected to V_{BB}, CLK_X provides the controlling data clock. When FS_X is used as a transmit frame synchronization input, CLK_R provides the controlling data clock. FS_R always provides the receive frame synchronization input. When FS_R is in high-state, serial data is shifted into the input register on the falling edge of the appropriate data clock.

The digital input should conform to the A-law companding law. Table 1 lists the required digital codes and the D/A converter's normalized decode values. Figure 6 shows a typical set of companding curves.

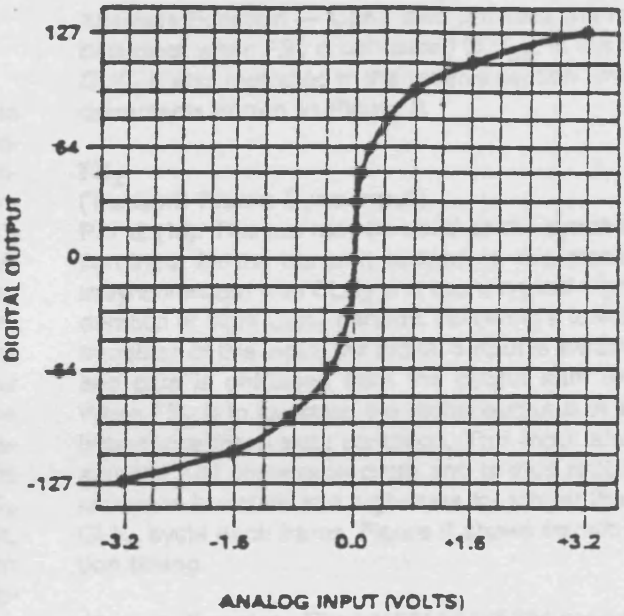
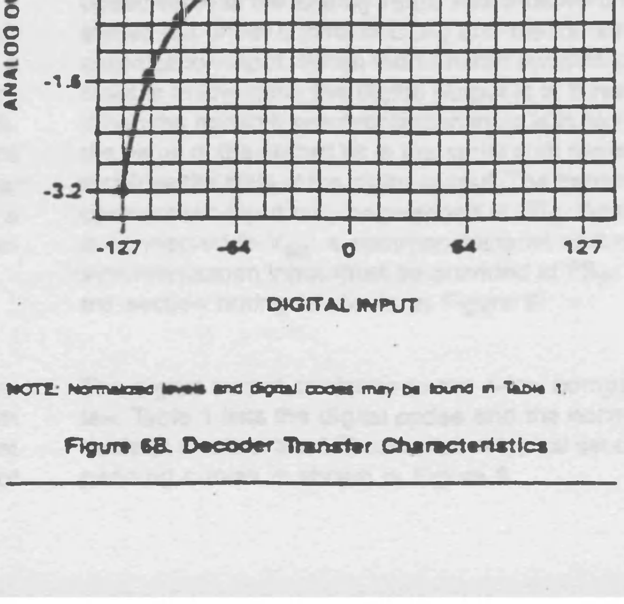


Figure 6A. Encoder Transfer Characteristics



NOTE: Normalized levels and digital codes may be found in Table 1.

Figure 6B. Decoder Transfer Characteristics

GRDD
(Digital Ground Input)
Pin 8 (6). This pin provides the ground return for the digital signals of the device. The digital ground is not internally connected to the analog ground. The analog ground and digital ground should be connected near the system supply ground.

FS_R
(Receive Frame Sync Input)
Pin 9 (7). This pin provides the synchronization input for the receive section. This input should be synchronized with the receive data clock and has a typical high-state duration of approximately eight receive clock periods. Data transfer occurs on the receive data clock's falling edge when FS_R is in high-state. The negative edge of this input should occur after the data clock has registered eight successive valid data bits. The D/A conversion using the last eight bits of serial data begins when FS_R completes a high-to-low transition. FS_R must remain in low-state and high-state for longer than one CLK_R period each frame to insure that the D/A command is acknowledged. When FS_X is connected to V_{BB}, CLK_X provides the receive data clock. When FS_X is used as a transmit frame synchronization input, CLK_R provides the receive data clock. Receive section timing is shown in Figure 8. FS_R also provides the synchronization input for controlling the transmit register when FS_X is connected to V_{BB}. In this mode, FS_R is also restricted to the transmit section timing requirements shown in Figure 9.

Alternate Function - This pin also provides a power standby mode during which power dissipation is significantly reduced. If FS_R remains in low-state for several time frames, the device will enter Standby Mode using limited power until FS_R is restarted. For proper operation of this function, CLK_R must always be provided with a standard system clock, as specified under the timing specifications.

CLK_R
(Receive Data Clock Input)
Pin 10 (8). This pin provides the data clock for controlling the receive data register when FS_X is used as a transmit frame synchronization input. When FS_X is connected to V_{BB}, CLK_X provides the receive data clock. In either mode, CLK_R must be provided with a standard system clock as specified under the timing specifications. CLK_R must be synchronous with FS_R, as a master clock used for internal timing and control is derived from these inputs.

CLK_X
(Transmit Data Clock Input)
Pin 11 (9). This pin provides the data clock input for controlling the transmit data register. Data is unloaded from the 8-bit output shift register at the clock rate present on this pin. Data transfer occurs on the rising edge of CLK_X when the transmit synchronization input is in high-state. The transmit synchronization input may be provided at FS_X. When FS_X is connected to V_{BB}, a common transmit and receive synchronization input must be provided at FS_R. Transmit section timing is shown in Figure 9.

Alternate Function — CLK_X also provides the receive data clock when FS_X is connected to V_{BB}. In this mode, CLK_X is also restricted to the receive section timing requirements shown in Figure 8.

FS_X
(Transmit Frame Sync Input)
Pin 12 (10). This pin may be used as the synchronization input for the transmit section. In this mode, FS_X is synchronized with CLK_X and has a typical high-state duration of eight CLK_X periods. Following a low-to-high transition of this input, the digital output is switched on and data is unloaded from the output shift register. When FS_X is in low-state, the digital output is in a high-impedance three-state condition. This input also initiates the A/D conversion cycle and is thus required to remain in low-state and high-state for longer than one CLK_R cycle each frame. Figure 9 shows transmit section timing.

Alternate Function - The MK5356/MK5326 provides an Alternate Register Control Mode by connecting FS_X to V_{BB}. In this mode, the transmit and receive sections operate synchronously. The CLK_X and FS_R inputs control both transmit and receive data registers. If FS_X is connected to V_{BB} with the power supplies connected, the MK5356/MK5326 should be cycled through the Power-Down or Standby Mode to insure alternate register control.

D_X
(Transmit Data Serial Output)
Pin 13 (11). This pin provides the digital output of the transmit data register. Upon completion of an A/D conversion, the transmit data register stores the 8-bit encoded value of the analog input. This 8-bit word is then shifted out under control of CLK_X and the transmit synchronization input. When the transmit synchronization input is in low-state, the digital output is in three-state. When the transmit synchronization input is in high-state, the value of the output bit in the serial shift register determines the state of the digital output. The transmit synchronization input may be provided at FS_X. When FS_X is connected to V_{BB}, a common transmit and receive synchronization input must be provided at FS_R. Transmit section timing is shown in Figure 9.

The digital output conforms to the A-law companding law. Table 1 lists the digital codes and the normalized decision levels of the A/D converter. A typical set of companding curves is shown in Figure 6.

)
›wer Down Input)
ı 14 (12). This pin provides the logic input for con-
lling the Active and Power-Down Modes. System de-
ıns typically operate the MK5356/MK5326 between
ı Power-Down Mode and the Active Mode. When the
r-state of PD is near GRDD, an Active Mode operat-
ı with A-law companding characteristics results. A
ıh-state places the device in the Power-Down Mode.

$_X I_C$
ıice-Frequency Analog Input)
ı 15 (13). This pin provides the analog input into the
:5356/MK5326 encoder. This input is typically ac-
ıpled from VF_XO_F, the transmit filter stage output.
ı encoder then samples and performs the A/D con-
sions on this band-limited voice frequency signal.
mpling is performed at a rate equal to the typical 8
ː rate of the transmit synchronization input. The ana-
input amplitude to the encoder should range within
±3.2 V on-chip references for accurate conversions.

ı encoder's companding characteristics are shown
igure 5. Table 1 lists the A/D converter's digital codes
l normalized decision levels.

$_XO_F$
ice-Frequency Analog Output)
16 (14). This pin provides the analog output of the
ısmit filter. This output should be ac-coupled to the
oder section. A capacitor of value 0.1 μF is typically
nected between VF_XI_C and VF_XO_F to provide the
coupling.

I+
ce-Frequency Analog Noninverting Input)
17. Pin 17 provides the noninverting input to the
smit gain-adjust amplifier.

I−
ce-Frequency Analog Inverting Input)
18 (15). This pin provides the inverting input to the
smit gain-adjust amplifier. In the MK5356, the am-
ɪr's noninverting input is internally connected to
og ground. In the MK5326, Pin 17 provides the
nverting input. A gain configuration can be im-
ıented by using the amplifier output provided by
. The signal from the operational amplifier then
ses through the transmit filter. The transmit filter
d-limits the signal to be encoded and provides 6 dB
dditional gain in the pass-band. Figure 7 shows the
sfer characteristics of the transmit filter.

Alternate Function — In the MK5356/MK5326, the gain-adjust amplifier may be bypassed by connecting VF_XI- to V_{BB}. In the MK5326, the VF_XI+ input should be connected to GRDA. The analog input should then be applied directly to the transmit filter at GS_X. For both the MK5356 and MK5326, if the amplifier bypass connections are made with the power supplies connected, the devices should be cycled through the Power-Down or Standby Mode to insure the gain-adjust amplifier has been disabled.

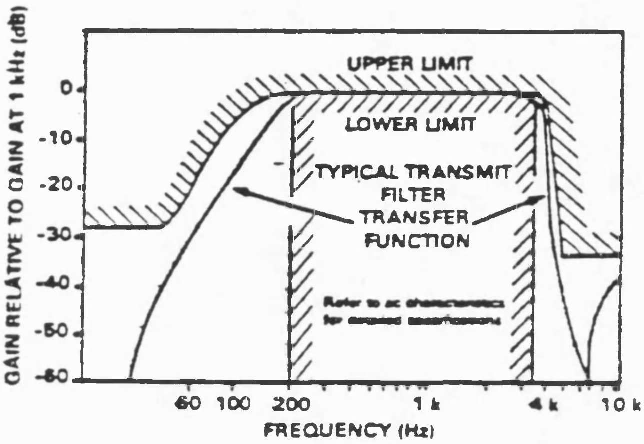


Figure 7. Transmit Filter Transfer Characteristics

GS_X
(Transmit Gain-Set Amplifier Output)
Pin 19 (16). This pin provides the output of the gain-adjust amplifier in the transmit filter section. The load connected to GS_X should not exceed a load of 10KΩ in parallel with 30 pF. The maximum output voltage range is ±1.7 V.

Alternate Function — GS_X can provide a direct input to the transmit filter with proper connections as described under the alternate function of VF_XI-.

GRDA
(Analog Ground Input)
Pin 20 (1). GRDA provides the ground return for the device's analog signals. The analog ground is not internally connected to the digital ground. The digital ground and analog ground should be connected near the system supply ground.

Table 1. MK5356/MK5326 A-Law Encode - Decode Characteristics

Chord Number	Number of Steps	Step Size	Normalized Encode Decision Levels	Digital Code 1	2	3	4	5	6	7	8	Normalized Decode Levels
			4096									
				1	0	1	0	1	0	1	0	4032
			3968									
7	16	128	:				:					:
			2176									
				1	0	1	0	0	1	0	1	2112
			2048									
6	16	64	:				:					:
			1088									
				1	0	1	1	0	1	0	1	1056
			1024									
5	16	32	:				:					:
			544									
				1	0	0	0	0	1	0	1	528
			512									
4	16	16	:				:					:
			272									
				1	0	0	1	0	1	0	1	264
			256									
3	16	8	:				:					:
			136									
				1	1	1	0	0	1	0	1	132
			128									
2	16	4	:				:					:
			68									
				1	1	1	1	0	1	0	1	66
			64									
1	32	2	:				:					:
			2									
				1	1	0	1	0	1	0	1	1
			0									

NOTES:
1. Characteristics are symmetrical about analog zero with sign bit = 0 for negative analog values.
2. Transmit and receive levels may be found under ac characteristics.

BSOLUTE MAXIMUM RATINGS*

ositive Supply Voltage .... GRDD $\leq V_{CC} \leq +6$ V
egative Supply Voltage .... -6 V $\leq V_{BB} \leq$ GRDD
igital Ground (GRDD) with respect to Analog Ground (GRDA) .... -250 mV $\leq$ GRDD $\leq$ 250 mV
l Other Inputs .... V_{BB} -0.3 V $\leq V_{IN} \leq V_{CC}$ $+0.3$V
ackage Dissipation at 25°C (Derate by 9 mW/°C when soldered into PCB) .... 500 mW
mbient Operating Temperature .... 0°C $\leq T_A \leq$ 70°C
torage Temperature .... -55°C $\leq T_S \leq$ 125°C

tresses above those listed under "Absolute Maximum Ratings" may cause permanent damage to the device. This is a stress rating only and functional operation the device at these or any other condition above those indicated in the operational sections of this specification is not implied. Exposure to absolute maximum ing conditions for extended periods may affect device reliability.

LECTRICAL SPECIFICATIONS

C CHARACTERISTICS

'$\leq T_A \leq$ 70°C, V_{CC} = +5 V ± 5%, V_{BB} = -5 V ± 5%, GRDA = 0 V, GRDD = 0 V unless otherwise specified

OWER SUPPLY

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
I_{CC}	Operating Current, V_{CC}		4.5	7	mAdc	Analog inputs at GRDA; outputs unloaded; power amps disabled; clocks applied.
I_{BB}	Operating Current, V_{BB}		3.5	6	mAdc	
P_{OP}	Operating Power Dissipation		40		mW	Note 1
I_{CCP}	Power Down Current, V_{CC}		0.1	2	μAdc	PD to V_{CC}; clocks applied.
I_{BBP}	Power Down Current, V_{BB}		0.1	2	μAdc	Note 1
P_{DP}	Power Down Power Dissipation		1		μW	
I_{CCS}	Standby Current, V_{CC}		150	500	μAdc	FS_R to GRDD; clocks applied
I_{BBS}	Standby Current, V_{BB}		0.1	2	μAdc	Note 1
P_{ST}	Standby Power Dissipation		0.75		mW	
I_{CCA}	Operating Current with Amplifiers Active, V_{CC}		5	8	mAdc	PWRI to GRDA; clocks applied
I_{BBA}	Operating Current with Amplifiers Active, V_{BB}		4	7	mAdc	Note 2
P_{AA}	Power Dissipation with Amplifiers Active		45		mW	

TES:
MK5356 and MK5326
MK5326 only

DIGITAL INTERFACE

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
V_{IL}	Input Low Voltage			0.6	Vdc	
V_{IH}	Input High Voltage	2.2			Vdc	
V_{IHPD}	Input High Voltage, PD	2.2			Vdc	PD, Power-Down Mode
V_{ILPD}	Input Low Voltage, PD	−0.3		+0.6	Vdc	PD, Active Mode
V_{IF}	Alternate Function Input Voltage, FS_X, VF_XI-, PWRI	V_{BB} −0.3		V_{BB} +0.6	Vdc	
I_I	Input Current, D_R, FS_R, CLK_R, CLK_X, FS_X	−10	±0.1	20	μAdc	$V_{BB} \leq V_{IN} \leq V_{CC}$
V_{OL}	Output Low Voltage, D_X			0.4	Vdc	I_{OL} = 1.6 mA
V_{OH}	Output High Voltage, D_X	2.4			Vdc	I_{OH} = −400 μA
		V_{CC} −1.0			Vdc	I_{OH} = −100 μA
C_{DO}	Digital Output Capacitance		5		pF	Digital output in three-state
I_{DOL}	Digital Output Leakage Current	−10	±0.1	+10	μAdc	Digital output in three-state

ANALOG INTERFACE, TRANSMIT FILTER AND ENCODER

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
I_{LX}	Input Leakage Current, G_{SX}	−100		+100	μA	−1.7 V ≤ V_{IN} ≤ 1.7 V; VF_XI- to V_{BB}; VF_XI+ to GRDA
R_{OX}	Output Resistance, VF_XO_F		1		Ω	
V_{OX}	Output Voltage Swing, VF_XO_F	±3.2			V	VF_XI- to V_{BB}; VF_XI+ to GRDA: Load 10 kΩ, 30 pF; 1 kHz at GS_X
C_{LX}	Load Capacitance, VF_XO_F			30	pF	
R_{LX}	Load Resistance, VF_XO_F	10			kΩ	
R_{IC}	Input Resistance, VF_XI_C	20	50		kΩ	
C_{ISC}	Input Sampling Capacitance, VF_XI_C		50		pF	
V_{OSC}	Encoder Offset Voltage, D_X	−5		+5	mV	VF_XI_C to GRDA thru 0.1 μF; Measure D_X

YM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
SRR1	Power Supply Rejection of V_{CC} at D_X 1 kHz 3 kHz		 35 30		 dB dB	Inverting Gain Input Amp; R_I = 10 kΩ; Injection Signal of 70 mV_{rms}; ±5 V supplies; Measure Narrow Band
SRR2	Power Supply Rejection of V_{BB} at D_X 1 kHz 3 kHz		 45 40		 dB dB	Inverting Gain Input Amp; R_I = 10 kΩ; Injection Signal of 70 mV_{rms}; ±5 V supplies; Measure Narrow Band

ALOG INTERFACE, TRANSMIT GAIN-SETTING AMPLIFIER (MK5356)

M	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
A	Input Leakage Current, VF_XI-	−200		+200	nA	$-1.7\ V \le V_{IN} \le 1.7\ V$
IA	Input Resistance, VF_XI-	10			MΩ	
SXI	Input Offset Voltage, VF_XI-	−25		+25	mV	
L	Open Loop Voltage Gain		5000		V/V	
	Open Loop Unity Gain Bandwidth		1		MHz	
IA	Output Voltage Swing, GS_X	±1.7			V	Load 10 Ω, 30 pF
IA	Load Capacitance, GS_X			30	pF	
A	Load Resistance, GS_X	10			kΩ	

LOG INTERFACE, TRANSMIT GAIN-SETTING AMPLIFIER (MK5326)

	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
	Input Leakage Current, VF_XI-, VF_XI+	−200		+200	nA	$-1.7\ V \le V_{IN} \le 1.7\ V$
	Input Resistance, VF_XI-, VF_XI+	10			MΩ	
A	Input Offset Voltage, VF_XI-, VF_XI+	−25		+25	mV	
R	Common Mode Rejection, VF_XI-, VF_XI+		70		dB	
	Open Loop Voltage Gain		5000		V/V	
	Open Loop Unity Gain Bandwidth		1		MHz	
	Output Voltage Swing, GS_X	±1.7			V	Load 10 kΩ, 30 pF
	Load Capacitance, GS_X			30	pF	
	Load Resistance, GS_X	10			kΩ	

ANALOG INTERFACE, RECEIVE FILTER

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
R_{OR}	Output Resistance, VF_RO		1		Ω	
V_{OSR}	Output Offset Voltage, VF_RO	−200		+200	mV	Zero Code at D_R
V_{OR}	Output Voltage Swing, VF_RO		±3.2		V	Load 10 kΩ, 30 pF
C_{LR}	Load Capacitance, VF_RO			30	pF	
R_{LR}	Load Resistance, VF_RO	10			kΩ	
PSRR3	Power Supply Rejection of V_{CC} at VF_RO 1 kHz 3 kHz		35 35		dB dB	Injection Signal of 70 mV_{rms}; ±5 V supplies; Measure Narrow Band
PSRR4	Power Supply Rejection V_{BB} at VF_RO 1 kHz 3 kHz		45 45		dB dB	Injection Signal of 70 mV_{rms}; ±5 V supplies; Measure Narrow Band

ANALOG INTERFACE, RECEIVE POWER AMPLIFIER STAGE (MK5326)

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
I_{LRA}	Input Leakage Current, PWRI	−320		+320	nA	Input Range ±3.2 V
R_{IRA}	Input Resistance, PWRI	10			MΩ	
G_{ARA}	Absolute Gain, PWRO+	−0.1		+0.1	dB	0 dBm0 at PWRI R_L = 300Ω to GRDA
G_{RRA}	Relative Gain, PWRO- to PWRO+	−0.1		+0.1	dB	0 dBm0 at PWRI R_L = 300Ω to GRDA
f_{CR}	Unity Gain Bandwidth, PWRO+, PWRO−		1		MHz	
R_{ORA}	Output Resistance, PWRO+ PWRO−		1	4	Ω	Output Range ±3.2 V
V_{OSRA}	Output Offset Voltage, PWRO+, PWRO−	−50		+50	mV	PWRI to GRDA R_L = 300Ω to GRDA
C_{LRA}	Load Capacitance, PWRO+, PWRO−			200	pF	
V_{ORA1}	Single-Ended Output Voltage Swing, PWRO+, PWRO−	±3.2			V	R_L = 300 Ω to GRDA; ±5 V supplies
V_{ORA2}	Differential Output Voltage Swing, PWRO+ to PWRO−	±6.4			V	R_L = 600 Ω from PWRO+ to PWRO−; ±5 V supplies
SD_{RA}	Signal to Distortion, PWRO+, PWRO−	60	70		dBp	0 dBm0 at PWRI

AC CHARACTERISTICS

$°C \le T_A \le 70°C$, V_{CC} = +5 V ±5%, V_{BB} = −5 V ±5%, GRDA = 0 V, GRDD = 0 V unless otherwise specified.

TRANSMIT FILTER AND ENCODER CHARACTERISTICS

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
G_{RX}	Relative Gain, D_X				dB	Inverting Gain Input Amp; R_f = 10 kΩ; 0 dBm0 Input Level; Reference 820 Hz
	50 Hz and Below			−30		
	60 Hz			−26		
	200 Hz	−2.2		−.10		
	300 Hz to 3000 Hz	−0.15		+0.15		
	3300 Hz	−0.35		+.05		
	3400 Hz	−0.7		0.00		
	3600 Hz			−1.7		
	4000 Hz			−14.0		
	4600 Hz and Above			−33.0	dB	Measure 0 to 4 kHz
G_{AF}	Absolute Gain, Transmit Filter, VF_XO_F		6.0		dB	±5 V Supplies 820 Hz
G_{AX}	Absolute Gain Transmit, D_X	5.80	6.0	6.20	dB	820 Hz at GS_X ; ±5% Supplies; $0°C \le T_A \le 70°C$
$SD1_X$	Transmit Signal to Distortion, D_X	37 31 26			dBp	0 to −30 dBm0 −40 dBm0 −45 dBm0 Inverting Gain Input Amp; R_f = 10 kΩ; 820 Hz
$SD2_X$	Transmit Signal to Distortion, D_X	30 36 34 30 15			dB	-3 dBm0 -6 to -27 dBm0 -34 dBm0 -40 dBm0 -55 dBm0 Inverting Gain Input Amp; R_f = 10 kΩ; Narrow Band Noise Input
SFD_X	Single Frequency Distortion Products, D_X			−46	dB	Inverting Gain Input Amp; R_f = 10 kΩ; Input 0 dBm0 at 820 Hz
T_X	Transmit Gain Tracking, D_X	−0.2 −0.4 -1.25	±0.1 ±0.1 ±0.2	+0.2 +0.4 +1.25	dB	+3 to −40 dBm0 −40 to −50 dBm0 -50 to -55 dBm0 Inverting Gain Input Amp; R_f = 10 kΩ; Input at 820 Hz; -10 dBm0 reference signal
$_X$	Transmit Idle Channel Noise, D_X			−68	dBm0p	Inverting Gain Input Amp; R_f = 10 kΩ; Input to GRDA
$_{DX}$	Differential Envelope Delay, VF_XO_F		40		µs	Inverting Gain Input Amp; R_f = 10 kΩ; Input 0 dBm0; 1000 Hz to 2600 Hz
$_{AX}$	Absolute Delay, VF_XO_F		195		µs	Input 0 dBm0 at 1000 Hz

DECODER AND RECEIVE FILTER CHARACTERISTICS

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
G_{RR}	Relative Gain, VF_RO				dB	0 dBm0 Input Level; Reference 820 Hz
	200 Hz and Below			+.15		
	300 to 3000 Hz	−.15		+.15		
	3300 Hz	−0.35		+.05		
	3400 Hz	−0.7		0.0		
	3600 Hz			−1.7		
	4000 Hz			−14.0		
G_{AR}	Absolute Gain Receive, VF_RO	−.20	0.0	+.20	dB	820 Hz at D_R ; ±5% Supplies; $0°C \le T_A \le 70°C$
SS_{OB}	Spurious Out-of-Band Signals; 4600 Hz and Above, VF_RO			−40	dB	0 dBm0 Input at 820 Hz
$SD1_R$	Receive Signal to Distortion, VF_RO	37 31 26			dBp	0 to −30 dBm0 −40 dBm0 −45 dBm0 820 Hz Input at D_R
$SD2_R$	Receive Signal to Distortion, VF_R0	30 37 35 31 16			dB	-3 dBm0 -6 to −27 dBm0 -34 dBm0 -40 dBm0 -55 dBm0 Narrow Band Noise Input at D_R
SFD_R	Single Frequency Distortion Products, VF_RO			−46	dB	0 dBm0 at 820 Hz; Input at D_R
GT_R	Receive Gain Tracking, VF_RO	−0.2 −0.4 -1.25	±0.1 ±0.1 ±0.2	+0.2 +0.4 +1.25	dB	+3 to −40 dBm0 −40 to −50 dBm0 -50 to -55 dBm0 820 Hz Input at D_R
N_R	Receive Idle Channel Noise, VF_RO			-79	dBm0p	Positive Zero Code at D_R
D_{DR}	Differential Envelope Delay, VF_RO		120		μs	Input 0 dBm0; 1000 Hz to 2600 Hz
D_{AR}	Absolute Delay, VF_RO		125		μs	Input 0 dBm0 at 1000 Hz

SYSTEM-RELATED CHARACTERISTICS

SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
G_{REE}	Relative Gain, End-to-End VF_RO				dB	Inverting Gain Input Amp; R_f = 10 kΩ; 0 dBm0 Input Level; Reference 820 Hz
	50 Hz and Below		−30			
	60 Hz		−28			
	200 Hz		−1.0			
	300 Hz to 3000 Hz		±0.1			
	3300 Hz		−0.2			
	3400 Hz		−0.8			
	3600 Hz		−6.0			
	4000 Hz		−32			
	4600 Hz and Above		−40			
G_{AEE}	Absolute Gain End-to-End, VF_RO	5.8	6.0	6.2	dB	820 Hz at G_{SX}, ±5% Supplies $0°C \le T_A \le 70°C$
SD_{EE}	End-to-End Signal-to-Distortion, VF_RO	35 29 24			dBp	0 to −30 dBm0 −40 dBm0 −45 dBm0 Inverting Gain Input Amp; R_f = 10 kΩ; 820 Hz
SFD_{EE}	Single Frequency Distortion Products, End-to-End, VF_RO			−44	dB	Input 820 Hz; Inverting Gain Input Amp; R_f = 10 kΩ; Input 0 dBm0
GT_{EE}	End-to-End Gain Tracking, VF_RO	−0.4 −0.8 -2.5	±0.1 ±0.1 ±0.2	+0.4 +0.8 +2.5	dB	+3 to −40 dBm0 −40 to −50 dBm0 -50 to -55 dBm0 Inverting Gain Input Amp; R_f = 10 kΩ; 820 Hz
CT_{XR}	Transmit-to-Receive Crosstalk, VF_RO			−75	dB	Inverting Gain Input Amp; R_f =10 kΩ; Input 0 dBm0 at 820 Hz; Smallest Positive Zero Code at D_R
CT_{RX}	Receive-to-Transmit Crosstalk, D_X			−75	dB	Inverting Gain Input Amp; R_f = 10 kΩ; Input −50 dBm0 at 2600 Hz, Analog; Input 0 dBm0 820 Hz, D_R
LO_X	Transmit Overload Level, D_X		3.16		Vdc	
LO_R	Receive Overload Level, VF_RO		3.16		Vdc	
TLP	Transmission Level Point		0 6 6 6		dB	GS_X VF_XO_F VF_XI_C VF_RO

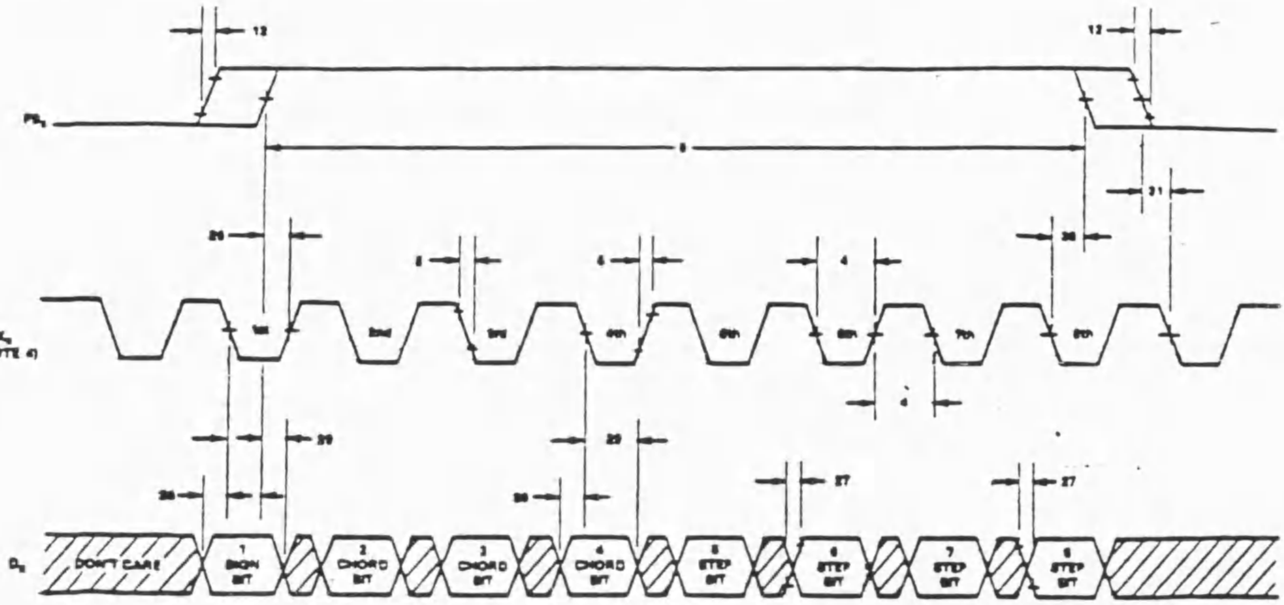
TIMING SPECIFICATIONS AND REQUIREMENTS

#	SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
1	t_{PD}	Power Delay Time			25	ms	On/Off Delay
2	f_{CX}	CLK_X Frequency	0.064		4.1	MHz	Notes 1, 2
3	f_{CF}	CLK_R Frequency		256 512 768 1024 1536 1544 2048 3072 4096		kHz	±0.1% of Typical Discrete Frequencies; Synchronized with FS_R Note 1
4	t_{CPH} t_{CPL}	Clock Pulse High and Low Times	40		60	% duty cycle	
5	t_{CR} t_{CF}	Clock Rise and Fall Times			100 50 25	ns	f_X, $f_R < 1.5$ MHz 1.5 MHz to 2.1 MHz f_X, $f_R > 2.1$ MHz
6	f_{SX}	FS_X Frequency		8		kHz	Notes 2, 3
7	f_{SR}	FS_R Frequency		8		kHz	±0.1%; Synchronized with CLK_R Notes 2, 3
8	t_{XSH}	FS_X High-State Time		$8/f_{CX}$		CLK_R periods	Note 3
9	t_{RSH}	FS_R High-State Time		$8/f_{CR}$		CLK_R periods	Note 3
10	t_{XSL}	FS_X Low-State Time	1.1			CLK_R periods	Note 3
11	t_{RSL}	FS_R Low-State Time	1.1			CLK_R periods	Note 3
12	t_{SR} t_{SF}	Sync Rise and Fall Times			100 50 25	ns	f_X, $f_R < 1.5$ MHz 1.5 MHz to 2.1 MHz f_X, $f_R > 2.1$ MHz
13	t_{XCS}	Transmit Clock-to-Sync Delay	25			ns	Notes 4, 5
14	t_{XSS}	Transmit Sync Set-Up Time	125			ns	Note 5

#	SYM	PARAMETER	MIN	TYP	MAX	UNITS	TEST CONDITIONS
15	t_{XDP}	Transmit Data Present Time	0		85	ns	$C_L = 30$ pF; $I_{OL} = -400\ \mu A$; $I_{OH} = -100\ \mu A$; (LS-TTL)
16	t_{XDD}	Transmit Data Delay Time	0		85	ns	
17	t_{XDDS}	Transmit Data Delay-From-Sync Time	0		100	ns	
18	t_{DOR} t_{DOF}	Digital Output Rise and Fall Times			50	ns	
19	t_{XDT}	Transmit Data Three-State Time			85	ns	Note 5
20	t_{XDP}	Transmit Data Present Time	0		150	ns	$C_L = 100$ pF; $I_{OL} = 1.6$ mA; $I_{OH} = -400\ \mu A$;
21	t_{XDD}	Transmit Data Delay Time	0		150	ns	
22	t_{XDDS}	Transmit Data Delay-From-Sync Time	0		175	ns	
23	t_{DOR} t_{DOF}	Digital Output Rise and Fall Times			100	ns	
24	t_{XDT}	Transmit Data Three-State Time			100	ns	Note 5
25	t_{XCSN}	Transmit Clock-to-Sync-Negative Edge Delay	85			ns	Note 5
26	t_{RSS}	Receive Sync Set-Up Time	75			ns	Note 5
27	t_{DIR} t_{DIF}	Digital Input Rise and Fall Times			100 50 25	ns	$f_X, f_R < 1.5$ MHz 1.5 MHz $< f_X, f_R < 2.1$ MHz $f_X, f_R > 2.1$ MHz
28	t_{RDS}	Receive Data Set-Up Time	25			ns	Note 5
29	t_{RDH}	Receive Data Hold Time	50			ns	Note 5
30	t_{RCSN}	Receive Clock-to-Sync Negative Edge Delay	75			ns	Note 5
31	t_{SNRC}	Receive Sync-Negative Edge-to-Clock Delay	25			ns	Notes 4, 5

NOTES:

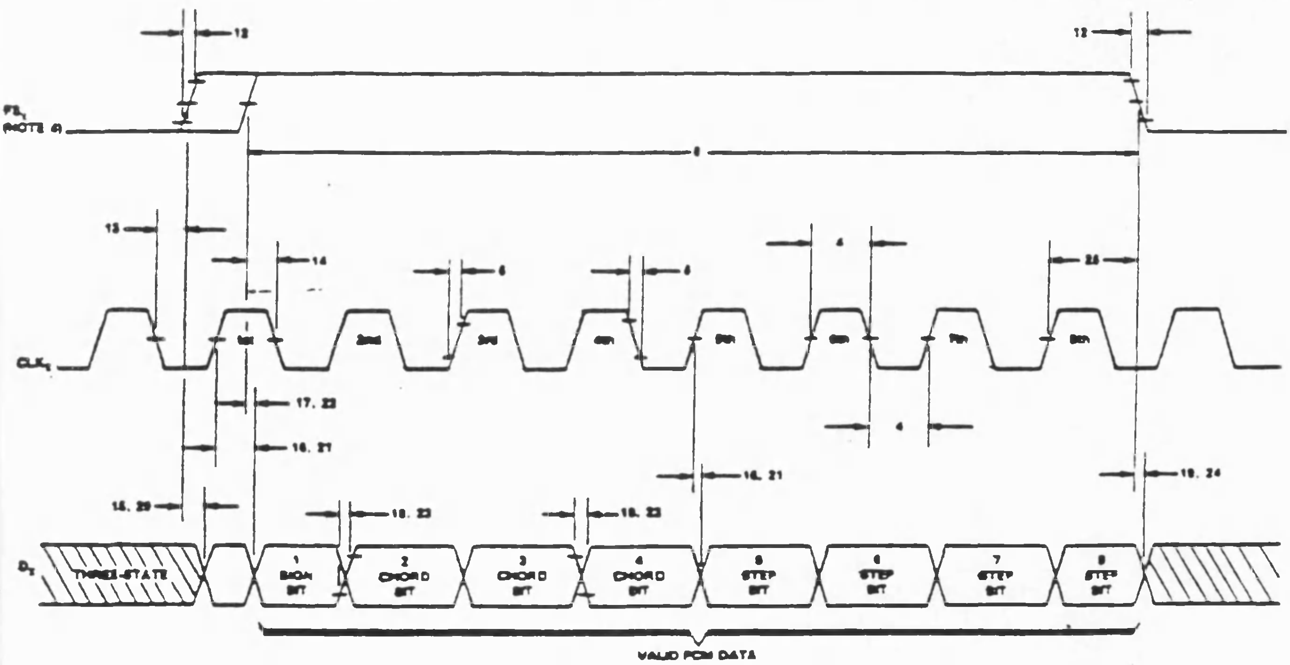
1. In the synchronous mode with FS_X tied to V_{BB}, both the transmit and receive data clocks are derived from CLK_X. CLK_X is never restricted to the discrete frequencies required at CLK_R. CLK_R is always restricted to the discrete frequencies listed under t_{CR}.
2. In the synchronous mode with FS_X tied to V_{BB}, the transmit and receive sections are controlled by a common data clock at CLK_X and a common frame sync at FS_R. In this mode CLK_X and FS_R must meet both the transmit and receive timing requirements of Figures 8 and 9.
3. Frame syncs are required to have a minimum high-state and low-state time greater than 1 CLK_R period time frame. This requirement insures that each change of state will be acknowledge internally.
4. This delay is necessary to avoid overlapping clock and sync.
5. Add 50% of the actual rise or fall time of the appropriate user input signal for parameters measured from 1.4 V trip levels.



NOTES:

1. For the first data bit, t_{RDS} and t_{RDH} should be measured from the rising edge of FS_R if the rising edge of FS_R occurs after the falling edge of the first CLK_R pulse.
2. System design may permit FS_R to be high longer than 8 CLK_R periods; the last 8 CLK_R rising edges will determine the 8 receive data bits.
3. Refer to the timing specification tables for t_{RSL}, a minimum low-state requirements for FS_R not shown in Figure 8.
4. The receive data clock signal is required at CLK_X when FS_X is tied to V_{BB}. In this mode, the transmit and receive sections are synchronous and CLK_X must meet both the transmit and receive section timing requirements.
5. Rise and fall times are measured using 0.4 V and 2.4 V levels. Delay times are measured from 1.4 V levels.

Figure 8. Receive Section Timing



NOTES:

1. For the first data bit, t_{XDDS} applies when the rising edge of FS_X occurs after the rising edge of the first CLK_X pulse, otherwise use t_{XDD}.
2. System design may permit FS_X to remain high longer than 8 CLK_X periods; the first 8 CLK_X rising edges will determine the 8 transmit data bits.
3. Refer to the timing specification tables for t_{XSL}, a minimum low state requirements for FS_X not shown in Figure 8.
4. The transmit frame synchronization signal is required at FS_R when FS_X is tied to V_{BB}. In this mode, the transmit and receive sections operate synchronously and FS_R must meet both the transmit and receive section timing requirements.
5. Rise and fall times are measured using 0.4 V and 2.4 V levels. Delay times are measured from 1.4 V levels.

Figure 9. Transmit Section Timing

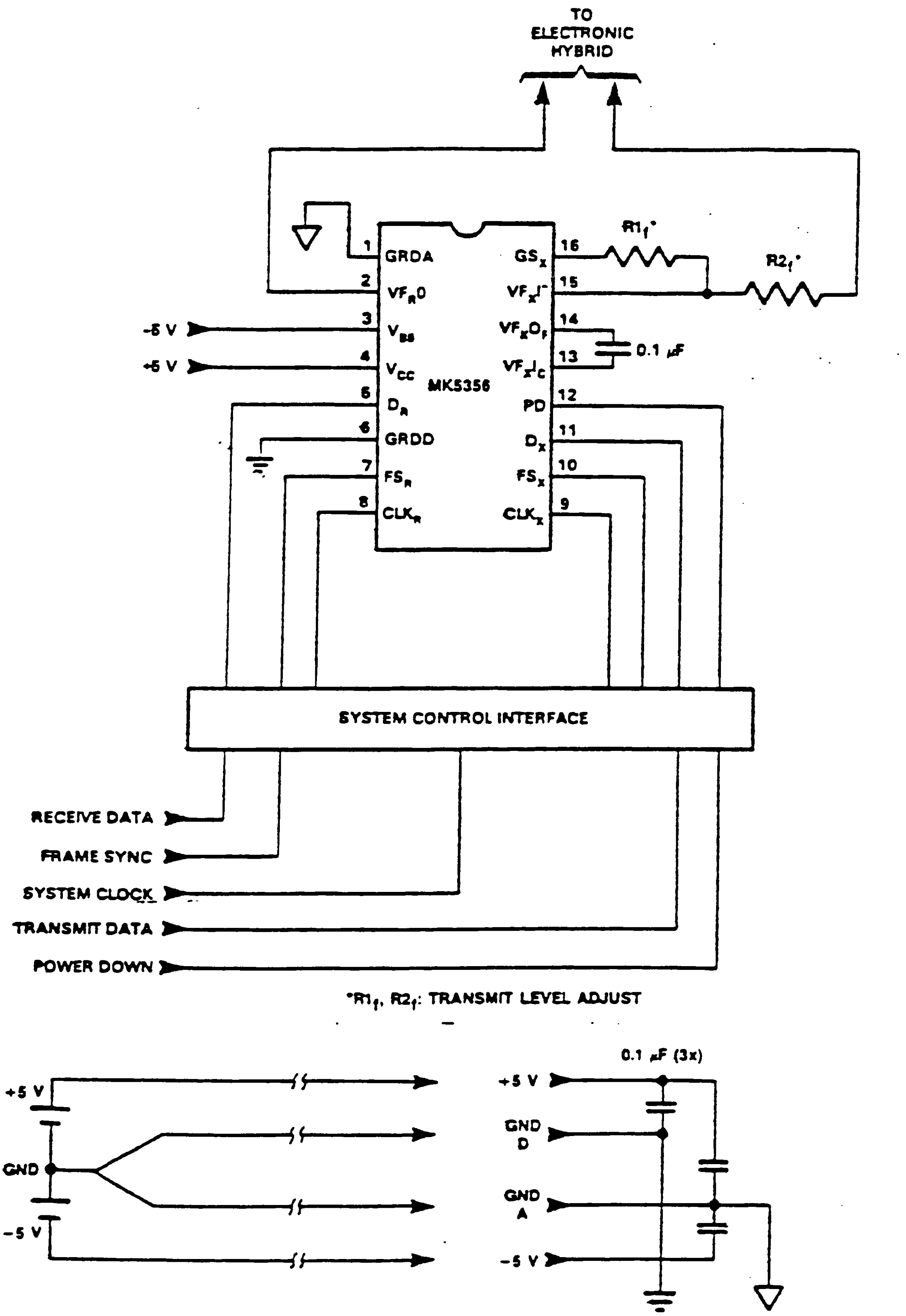


Figure 10. MK5356 Typical Application

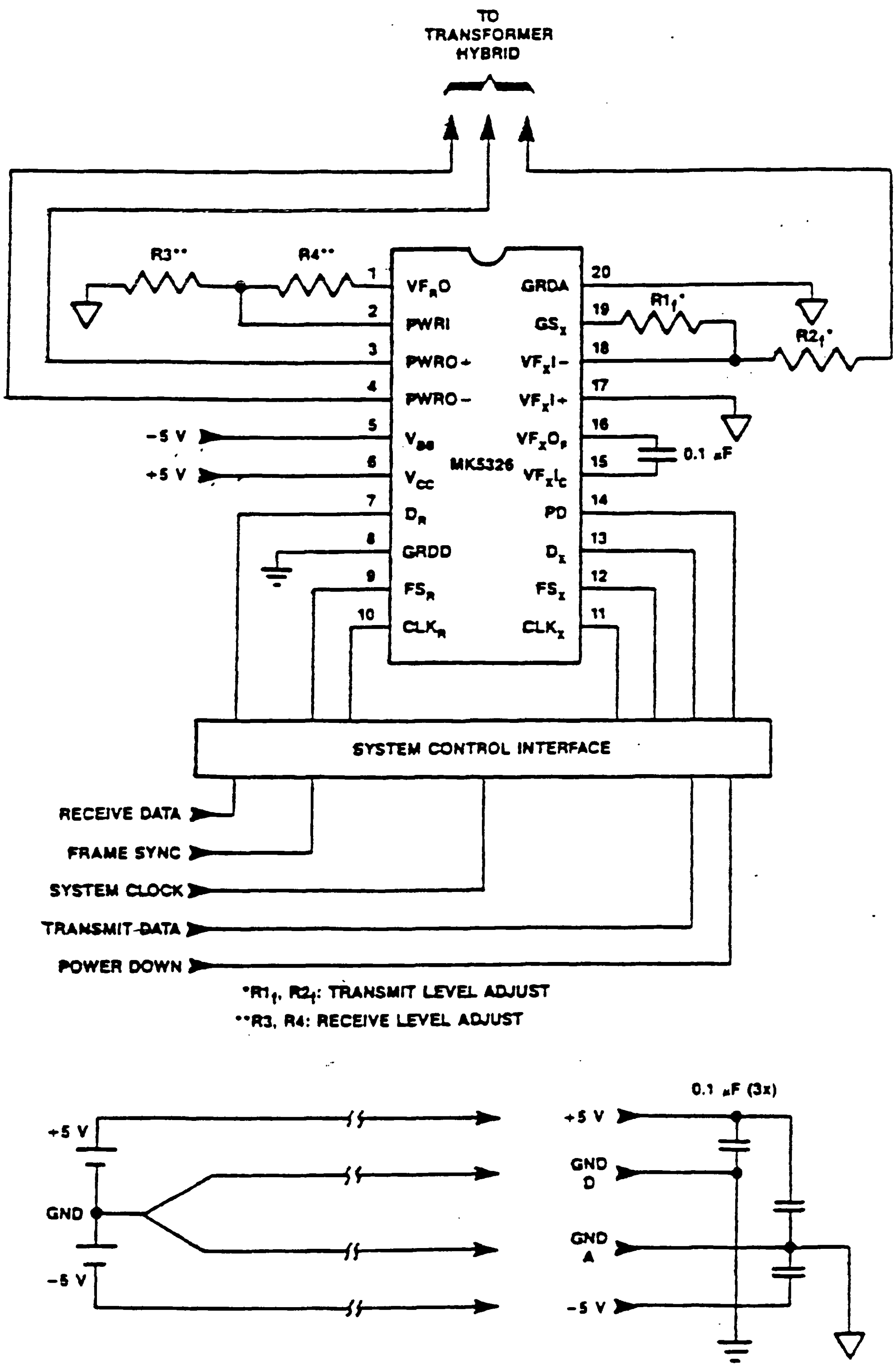


Figure 11. MK5326 Typical Application

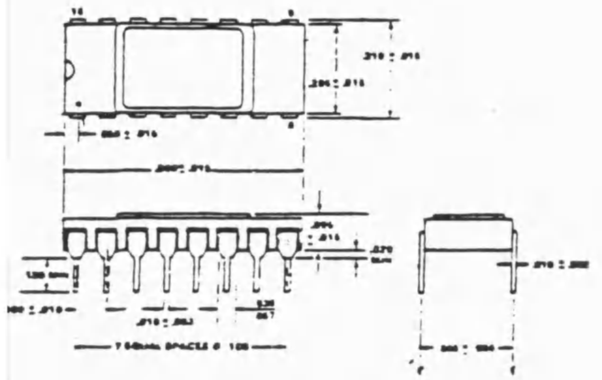


Figure 12. Package Description
Ceramic Dual-In-Line (P) 16-Pin MK5356(P)

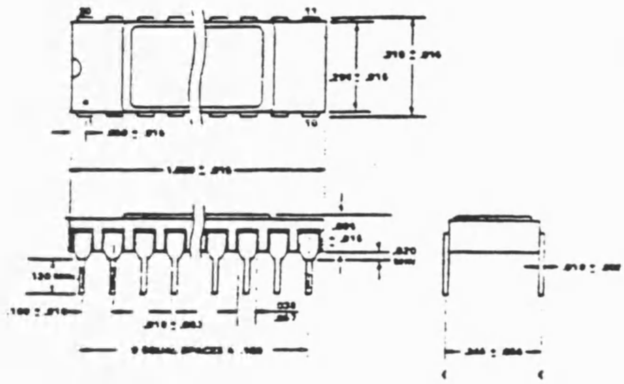


Figure 13. Package Description
Ceramic Dual-In-Line (P) 20-Pin MK5326(P)

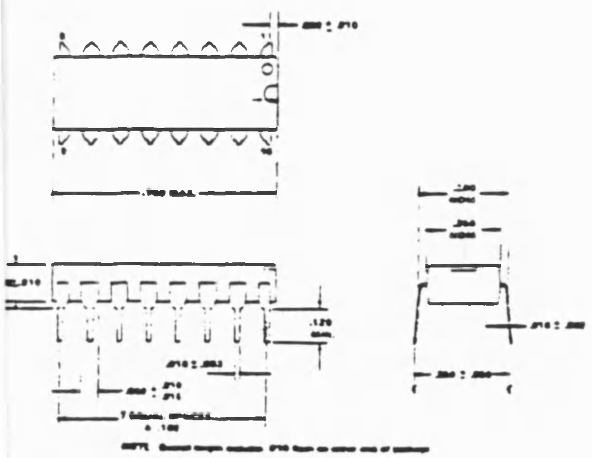


Figure 14. Package Description
Plastic Dual-In-Line (N) 16-Pin MK5356(N)

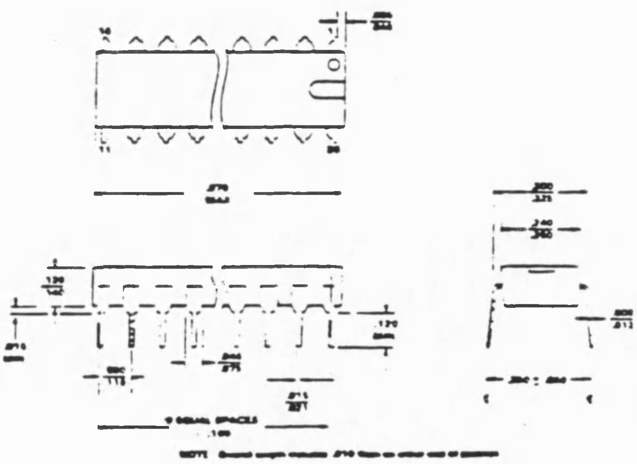


Figure 15. Package Description
Plastic Dual-In-Line (N) 20-Pin MK5326(N)

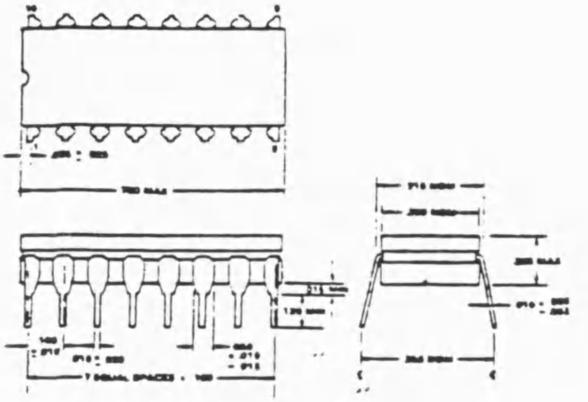


Figure 16. Package Description
Cerdip Hermetic (J) 16-Pin MK5356(J)

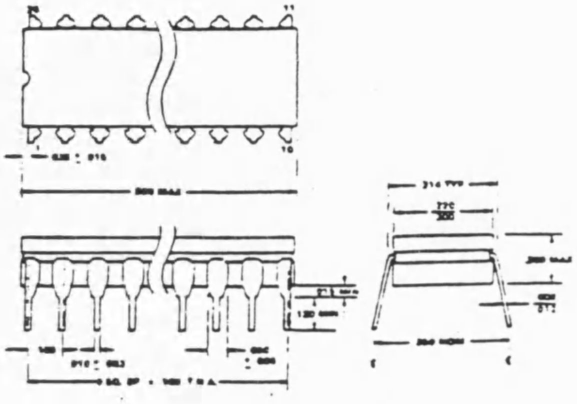
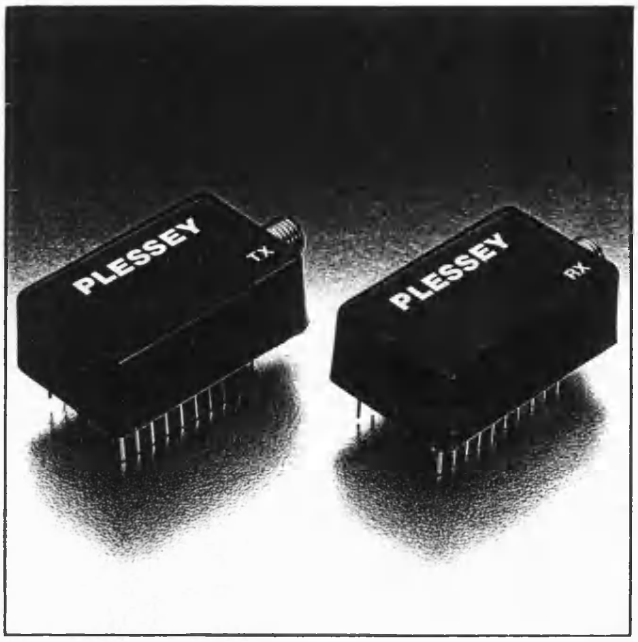


Figure 17. Package Description
Cerdip Hermetic (J) 20-Pin MK5326(J)

PLESSEY Opto ELECTRONICS

Fibre Optic Datalink

P35-8800



Applications

2–125 Mbit/s transmissions

Local area networking

Distances up to 2 km

Point to point data comms

Computer links

Inter-rack telecomms links

Features

Low cost

Manchester encoding up to 50 Mbit/s

NRZ transmission up to 125 Mbit/s (compatible with FDDI LAN standard)

ECL data transmission

Hermetically sealed ICs for high reliability

FMA optical interfaces

Product approval testing based on BS9450

Transmitter module

The transmitter module consists of a custom built IC and an LED housed in a 20-pin DIL package. FMA optical interfaces are included to allow flexibility in the choice of optical fibre.

Characteristics[1]

Parameter	Minimum	Typical	Maximum
Supply voltage	−4.95V	−5.2V	−5.45V
Supply current (I_{LED} = 80 mA)		160 mA	200 mA
ECL input levels	MECL 10K compatible		
Optical rise/fall times[2]	4 ns	6 ns	8 ns
Peak spectral emission	820 nm	850 nm	920 nm
Output power temperature dependence[3]		$-0.7\%°C^{-1}$	$-1.0\%°C^{-1}$
Data rate: NRZ mode	2 Mb/s		125 Mb/s
Data rate: Manchester mode	2 Mb/s		50 Mb/s

Optical power output

The minimum mean optical powers available vary with fibre used. These are indicated in the table below.

	Optical Output Power μW (dBm)	
Fibre diameter (μm)	**Minimum**	**Typical**
62.5/125	15 (−18.2)	25 (−16)
85/125	25 (−16)	45 (−13.5)
100/140	40 (−14)	70 (−11.5)

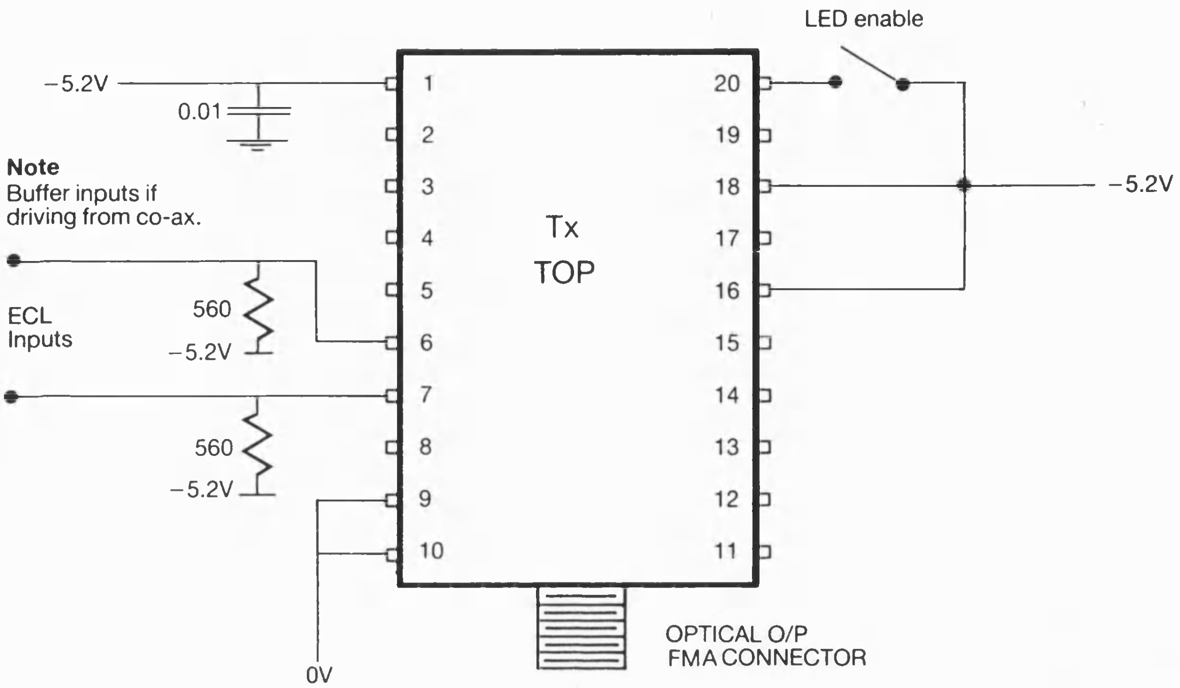
Notes

1. All performance is at 25°C ambient temp, $V_{EE} = -5.2V$ unless otherwise stated.
2. Measured across 10–90% levels.
3. Relative to output power at 25°C.

Limiting conditions of use

Operating temperature range	−5°C to +50°C
Storage temperature range	−20°C to +70°C
Maximum supply voltage	−7V

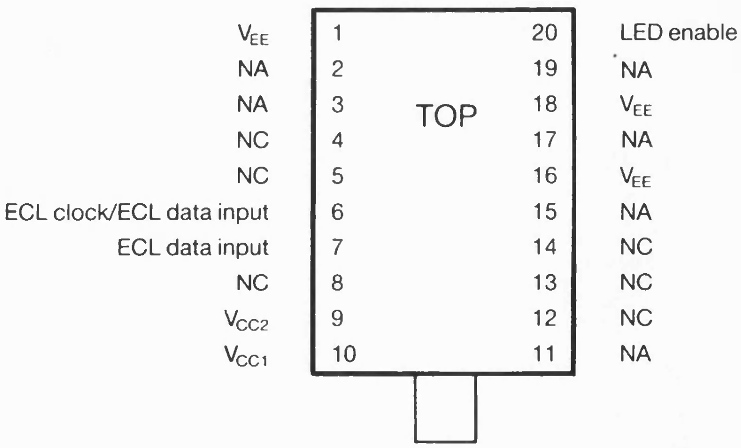
Transmitter electrical connections



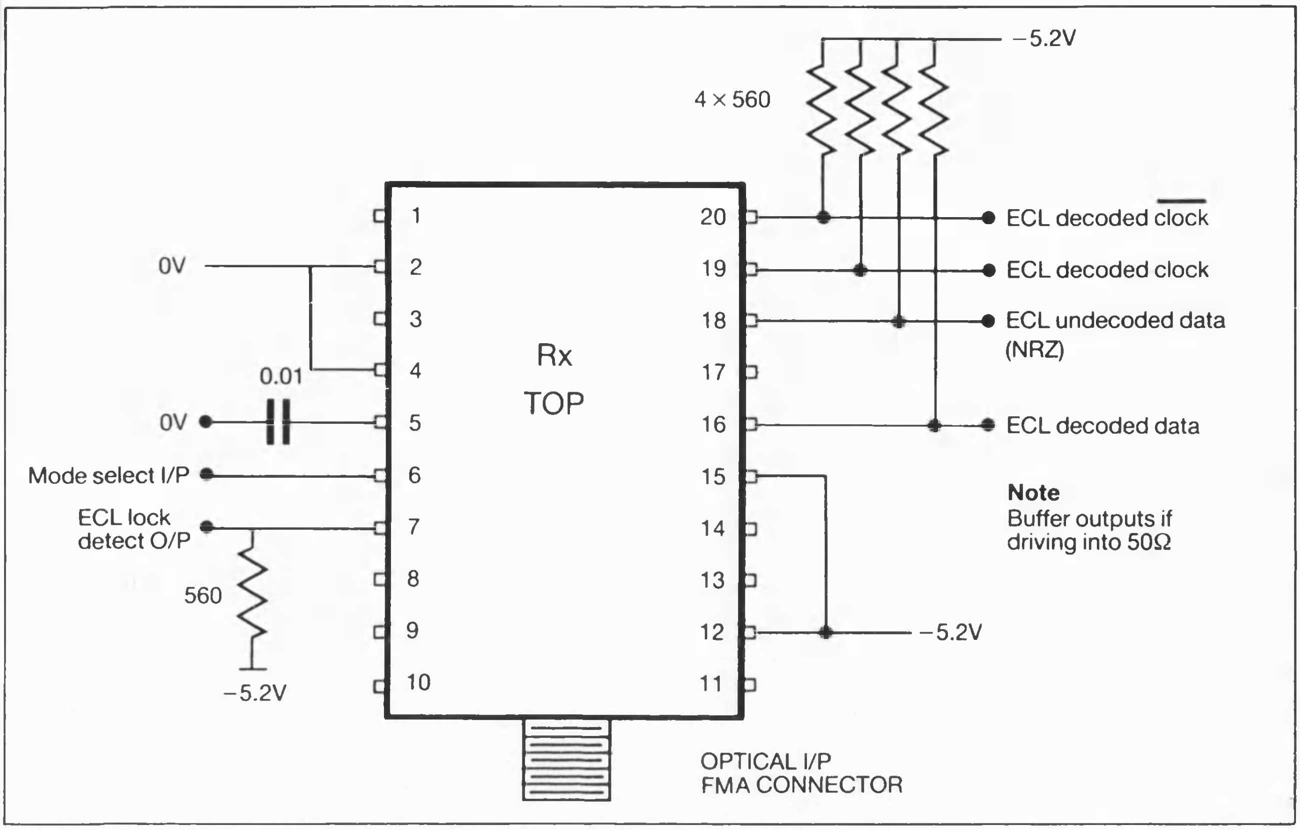
Operating mode	Pin 6	Pin 7
Manchester	Clock	Data
NRZ	Data	N/C

LED state	Pin 20
Enabled	O/C
Disabled	V_{EE}

Tx pin connections

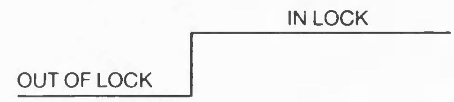


Receiver electrical connections



Operating mode	Pin 6
Manchester mode 1	V_{EE}
Manchester mode 2	Note 1
NRZ	O/C

Note 1
Connect to V_{EE} for 1μs, during start of preamble. Otherwise leave O/C.

ECL lock detect O/P


Note
Indicates decoded clock is frequency locked to incoming data (Manchester mode 2 only).

Rx pin connections

Function	Pin	Pin	Function
NC	1	20	ECL decoded $\overline{\text{clock}}$
V_{CC1}	2	19	ECL decoded clock
NC	3	18	ECL undecoded data (NRZ data)
V_{CC2}	4	17	NC
Loop filter capacitor	5	16	ECL decoded data
Mode select	6	15	V_{EE1}
ECL lock detect O/P	7	14	NA
NA	8	13	NA
NA	9	12	V_{EE2}
NA	10	11	NA

TOP

Receiver module

The receiving module is based on a custom built IC, incorporating a PIN photodiode and is, like the transmitter, housed in a 20-pin DIL package.

Characteristics

Parameter	Minimum	Typical	Maximum
Supply voltage V_{EE}	−4.95V	−5.2V	−5.45V
Supply current I_{CC}		150 mA	200 mA
Operating wavelength	820 nm	850 nm	920 nm
BER at rated sensitivity		10^{-10}	10^{-9}
Dynamic range	15 dB	17 dB	
ECL output load resistance	250Ω	470Ω	
ECL output rise/fall times		2 ns	4 ns
ECL output levels	MECL 10K compatible		

Receiver sensitivity

The minimum mean optical sensitivity for the Datalink receiver depends on the transmission option in use (Manchester or NRZ). The tables below indicate these differences.

NRZ option

Data rate*	2–100 Mb/s	Up to 125 Mb/s
Sensitivity	2μW (−27 dBm)	3μW (−25.2 dBm)

*Data must be balanced over a 1μs period

Manchester encoded

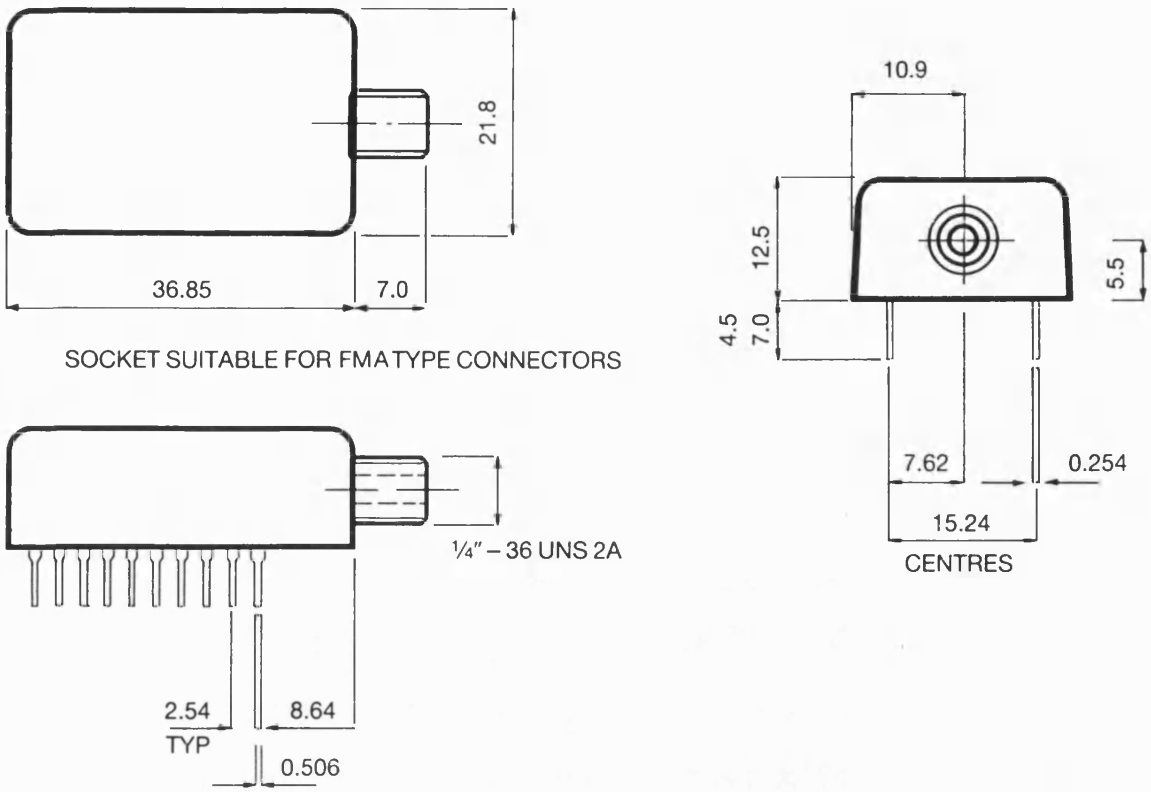
	Mode 1	Mode 2
Data rate	20–50 Mb/s	20–50 Mb/s
Sensitivity	2μW (−27 dBm)	2μW (−27 dBm)
Data string length maximum (continuous 1's or 0's)	2μs	Unlimited
Minimum start-up patterns (101010 . . .)	1μs	17.5μs

All performance data is at 25°C ambient temperature, V_{EE} = −5.2V unless otherwise stated.

Limiting conditions of use

Operating temperature	−5°C to +50°C
Storage temperature	−20°C to +70°C
Maximum supply voltage	−7V

Package outline – transmitter and receiver



Typical dimensions in mm

Important: Should further information be required, a full applications note is available on request.

PLESSEY®

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Publication No. 6123/1 86 Issue 1

Made in the United Kingdom

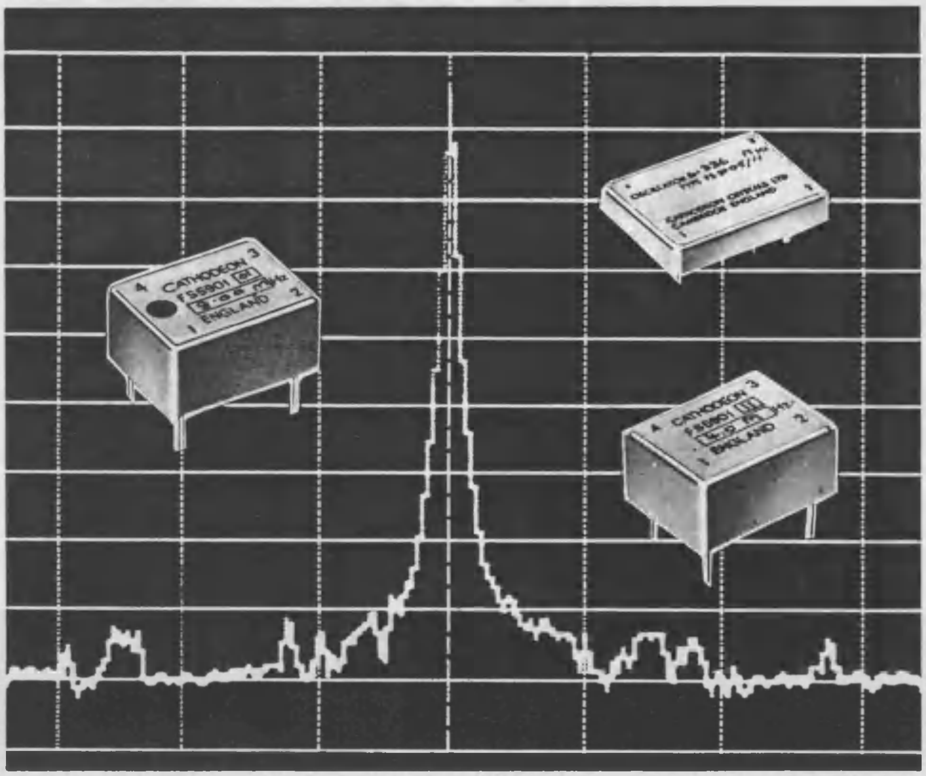
Crystal Oscillators

Cathodeon Crystals Ltd is approved to manufacture to BS 9600, BS 9610, BS 9620 and DEF-STAN 05-21

CONTENTS

This brochure offers a full range of Crystal Oscillators providing a total capability from design to production stages.

• Oven Controlled	(OCXO)
• Temperature Compensated	(TCXO)
• Voltage Controlled	(VCXO)
• Simple Packaged	(SPXO)
• General Purpose	(DIL)



GENERAL INFORMATION

Frequency Range

• OCXO	300 kHz – 40 MHz
• TCXO	5 MHz – 15 MHz
• SPXO	300 kHz – 40 MHz
• DIL	200 kHz – 70 MHz

How to Specify

- Order by specifying the oscillator type number with its appropriate suffix and the frequency, e.g.
  FS 5953–01 10.0 MHz
  Also specify the operating temperature range, frequency stability over the range, supply voltage and output waveform

Environmental Conditions

- As described in our Product Quality leaflet and in conformance with the recommended levels contained in BS 9620

New Products

This product brochure summarises the current range of Crystal Oscillators for general use. Enquiries are welcomed for types other than those published and should where possible include all relevant specification points, date required and potential quantity.

Construction

Most oscillators are manufactured in hermetically sealed cans with glass to metal seals for electrical termination. Dimensions are shown for each oscillator type on the back page.
High reliability components are used throughout the entire range and crystals are encapsulated in all-glass, cold weld or resistance weld holders to meet the specified ageing rate.
Full electrical, mechanical and environmental specifications are available on application.

CATHODEON
CRYSTALS

Cathodeon Crystals Ltd Linton Cambridge CB1 6JU England
Telephone 0223 891501 Telex 81212 Fax 0223 891955

Crystal Oscillator Products (STANDARD RANGE)

The Cathodeon standard range of Crystal Oscillators listed below is designed for a wide variety of applications where precise control of frequency, minimal power consumption and short warm-up time are essential.

High reliability components are used throughout the entire range with control of quality being monitored at each stage of manufacture. Computer optimisation is used for temperature compensated devices. Frequency versus temperature characteristics are controlled and determined by the crystal used and its associated circuitry.

All oscillators are tested by a computer controlled A.T.E., to assure the highest performance; test data can be provided on request.

Performance details are summarised in the table below.

OCXO Oven Controlled Crystal Oscillators are devices in which temperature sensitive components are maintained at a temperature above the maximum operating temperature, the crystal being designed to give optimum performance at this temperature. As stabilities of 0.1 ppm are readily achievable these devices are normally specified for applications requiring stabilities less than 0.5 ppm this being the limit of stability of a TCXO of comparable cost.

TCXO Temperature Compensated Crystal Oscillators are devices in which the frequency stability is achieved by compensating the temperature coefficient of the crystal using a temperature sensitive network. Each device is individually compensated using an advanced computer optimisation technique to provide the required performance with the minimum network complexity.

The major advantage of TCXO's is the relatively low power consumption and the fact that at switch on, regardless of the ambient temperature, the frequency is as specified. TCXO's are available with various output waveforms to suit the customers' requirements.

SPXO Simple Packaged Crystal Oscillators are devices in which the crystal is packaged with a maintaining circuit so as to provide the designer with a ready made frequency source.

VCXO Voltage Controlled Crystal Oscillators are SPXO in which the frequency can be controlled by the application of an external voltage. The devices are especially useful in phase locked loops. Up to ±100 ppm pulling can be achieved but ±50 ppm is more typical.

DIL Dual-in-Line Crystal Oscillators are miniature SPXO's in which a thick film containing the oscillator maintaining circuit and the crystal, are encapsulated in a hermetic enclosure, pin positions of which are the same as the standard DIL package.

	OCXO		TCXO		VCXO	SPXO				DIL
Type No.	FS 5951	FS 5953	FS 5805	FS 5815	FS 5909	FS 5901	FS 5903	FS 5905	FS 5906	FS 5911 FS 5912
Freq. Range	300 KHz to 40 MHz	300 KHz to 20 MHz	5 MHz to 15 MHz	5 MHz to 15 MHz	5 MHz to 20 MHz	300 KHz to 40 MHz	300 KHz to 40 MHz	4 MHz to 20 MHz	5 MHz to 20 MHz	250 KHz to 70 MHz
Output	TTL	TTL	TTL or Sine	TTL or Sine	TTL	TTL	TTL	TTL	TTL	TTL
Temp Range (°C) & Stability	0 to 60 ±0.1 ppm or −40 to 70 ±0.2 ppm	0 to 60 ±0.1 ppm or −40 to 70 ±0.2 ppm	−10 to 55 ±1 ppm or −20 to 70 ±2 ppm	−10 to 55 ±1 ppm or −20 to 70 ±2 ppm	−25 to 80 ±50 ppm on nominal	0 to 60 ±7.5 ppm or −40 to 80 ±25 ppm	0 to 60 ±7.5 ppm or −40 to 80 ±25 ppm	0 to 60 ±7.5 ppm or −40 to 80 ±25 ppm	0 to 60 ±7.5 ppm or −40 to 80 ±25 ppm	0 to 70 ±25 ppm on nominal
Freq. Adjusting	Internal Trimmer	Internal Trimmer & Ext. Volt.	External Resistor	External Potentiometer	External Volt. (50 ppm)	Internal Trimmer	Internal Trimmer & Ext. Volt.	—	Ext. Volt.	—
Ageing Rate	3×10^{-9} /day	3×10^{-9} /day	1 ppm/year	1 ppm/year	2 ppm/year	2 ppm/year	2 ppm/year	2ppm/year	2 ppm/year	Incl. in Temp. Tol.
Supply Osc.	5V	5V	9V	9V	5V to 15V	5V	5V	5V	5V	5V
Supply Oven	9–24V	9–24V								
Power or current consump. Osc.	40/60 mA	40/60 mA	9 mA	9 mA	20 mA at 12V	60 mA	60 mA	60 mA	60 mA	25 mA and 40 mA
Power or current consump. Oven	3–8W	3–8W								
Package	A	A	B	C	C	D	D	E	E	F

FREQUENCY TEMPERATURE PERFORMANCE

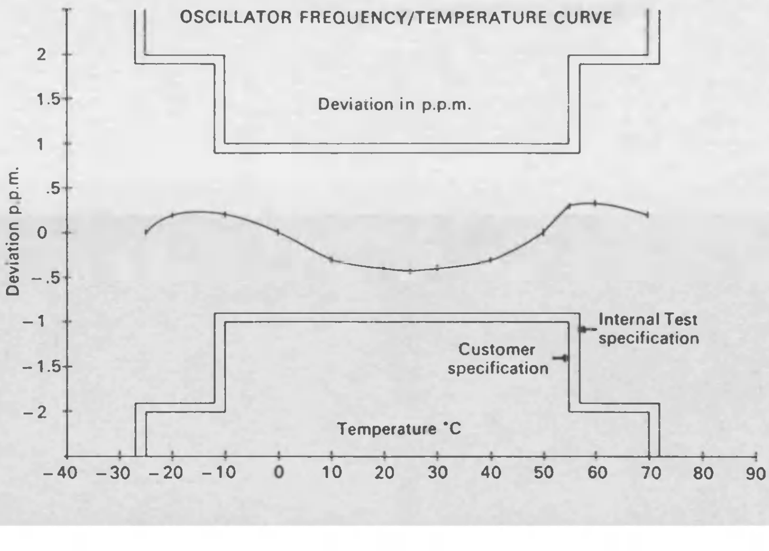
A wide variety of specifications are available for all oscillator types. Some typical specifications are listed.

For example OCXO's type FS 5951 are available to the following specifications:–

±0.1 ppm 0° to 60°C
or ±0.2 ppm −40°C to 70°C
or { ±0.1 ppm 0° to 60°C
±0.2 ppm −40° to 70°C }

plus other frequency/temperature combinations.

In general, the wider the temperature range the greater the cost due to extra testing.



)CXO Range

)VEN CONTROLLED CRYSTAL)SCILLATORS

athodeon Oven Controlled Crystal scillators (OCXO) provide a compact ighly stable frequency source suitable r use in frequency counters, fre-uency synthesisers and time bases.)ptions for remote adjustment, or hase locking of frequency are avail-ble through a varicap diode incorpor-ted in the oscillator.

his page defines the standard options f Cathodeon's Temperature Controlled rystal Oscillators in current produc-ion. The Company will consider manu-acture of variations on this and other ypes of oscillator.

he Phase Noise and Short Term tability of the FS 5951 and FS 5953 ange of oscillators is particularly good which makes them ideal for com-munications systems and microwave ources.

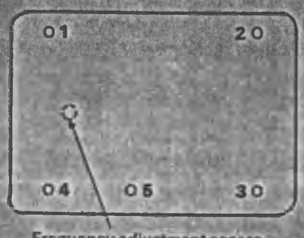
FS 5951 & FS 5953

)UTLINE SPECIFICATION

	FS 5951 SERIES		FS 5953 SERIES VOLTAGE CONTROLLED	
requency Range	Option – 03 – 01 – 04	300 kHz to 5 MHz 5 MHz to 20 MHz 20 MHz to 40 MHz	– 03 – 01	300 kHz to 5 MHz 5 MHz to 20 MHz
'ower Supply Options	Oscillator Oven	5V ±5% 8, 9, 12, 15, 18 or 24V	Oscillator Oven	5V ±5% 8, 9, 12, 15, 18 or 24V
'ower Consumption	Oscillator Option – 03 Other options Oven	 300mW max. 200mW max. 8W max. during warm-up period	Oscillator Option – 03 Other options Oven	 300mW max. 200mW max. 8W max. during warm-up period
Max. Operating Temperature Range	−40°C to 70°C		−40°C to 70°C	
Storage Temperature Range	−55°C to 105°C		−55°C to 105°C	
Tolerance of Frequency over Temperature Range Typical Specifications	±0.1 ppm ±0.2 ppm	0°C to 60°C −40°C to 70°C	±0.1 ppm ±0.2 ppm	0°C to 60°C −40°C to 70°C
Warm-up Time to ±0.3 ppm of Long Term Frequency	5 minutes		5 minutes	
RF Output Conditions	Option 03 Other options	TTL Fan-out 8 TTL Fan-out 2	Option 03 Other options	TTL Fan-out 8 TTL Fan-out 2
Typical Ageing Measured over a Period of 24 Hours	3×10^{-9} per day after 1 month		3×10^{-9} per day after 1 month	
Ageing Adjustment	Mechanical trimmer Range ±5 ppm min.		Mechanical trimmer Range ±5 ppm min.	
Voltage Control of Frequency	—		Voltage 1V to 10V. Sensitivity greater than 0.5 ppm per volt between 3V and 4V	

ELECTRICAL CONNECTIONS FS 5951 & FS 5953

View of base



1 Output
2 0 volts
3 +5 volts
4 Oven supply
5 Not connected – for FS 5951 series
Control voltage – for FS 5953 series

TCXO Range

EMPERATURE COMPENSATED RYSTAL OSCILLATORS

athodeon Crystals Temperature Compensated Crystal Oscillator (TCXO) is leally suited for applications requiring igh frequency stability, low power onsumption and immediate operation. he temperature range/frequency tability can normally be adjusted to uit customer requirements. The output an be sine wave or TTL compatible. eference should be made to the company for details.
he performance of these Oscillators as been closely matched to the requirements of the telecommunications market.

FS 5805 FS 5815

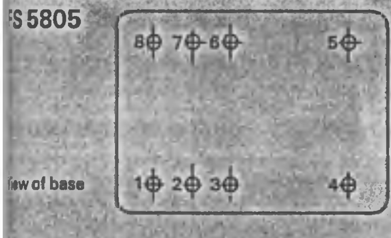
ELECTRICAL SPECIFICATIONS

UTLINE SPECIFICATION	FS 5805 SERIES	FS 5815 SERIES
requency Range	5 to 15 MHz	5 to 15 MHz
ower Supply Options	8, 9, 12, 15, 18, 20 and 24V	8, 9, 12, 15, 18, 20 and 24V
upply Current (maximum)	100 mW (Sine Wave O/P only)	100 mW (Sine Wave O/P only)
lax. Operating Temperature Range	−40°C to 85°C	−40°C to 85°C
torage Temperature Range	−55°C to 105°C	−55°C to 105°C
olerance of Frequency over emperature Range Typical specifications)	−40° to 85°C ±5 ppm 0° to 50°C ±0.5 ppm −20° to 70°C ±2 ppm { −25° to 70°C ±2 ppm −10° to 55°C ±1 ppm } 0° to 60°C ±2 ppm	−40° to 85°C ±5 ppm 0° to 50°C ±0.5 ppm −20° to 70°C ±2 ppm { −25° to 70°C ±2 ppm −10° to 55°C ±1 ppm } 0° to 60°C ±2 ppm
ate of Temperature Change	2°C per minute maximum	2°C per minute maximum
F Output Conditions	Option 00 – TTL compatible 01 – Open Collector TTL 02 – Sinusoidal	Option 00 – TTL compatible 01 – Open Collector TTL 02 – Sinusoidal
ypical Ageing Measured over a eriod of 24 Hours	Less than 1 ppm per year Less than 0.02 ppm per day	Less than 1 ppm per year Less than 0.02 ppm per day
geing Adjustment	By external connection of a 1 kΩ variable resistance	By external connection of a 10kΩ or 2k2Ω potentiometer—specify when ordering.

NOTE: A variety of options are available in respect of supply voltage, frequency vs temperature performance, package size and output. Full details together with electrical and toleranced mechanical specifications are available on request.

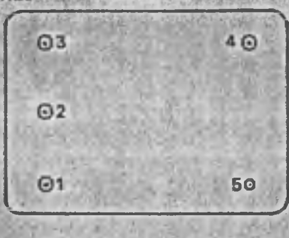
LECTRICAL CONNECTIONS

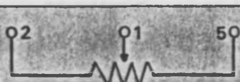
S 5805



1 Test Point – Do not use
2 Ageing Adjustment
3 Test Point – Do not use
4 R.F. Output
5 D.C. and R.F. Common
6 Test Point – Do not use
7 Test Point – Do not use
8 + D.C. Volts

FS 5815





1 Ageing Adjustment
2 Ageing Adjustment
3 + D.C. Volts
4 R.F. Output
5 D.C. and R.F. Common

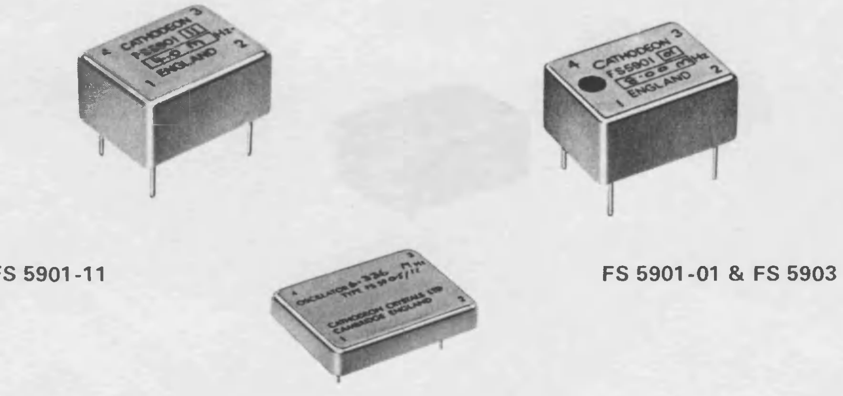
SPXO Range

SIMPLE PACKAGED CRYSTAL OSCILLATORS FS 5901, FS 5903, FS 5905 & FS 5906 SERIES

Cathodeon Crystals range of Simple Packaged Crystal Oscillators (SPXO) are designed to drive TTL logic. The range includes high stability and low profile versions. In some versions an option of remote frequency adjustment or facility for modulation or phase locking is available through a varicap diode.

This page defines some of the oscillators in current production. The company's capability extends far beyond this type of oscillator and consideration will be given to the development of other types.

The same basic drive circuit is used for this range of oscillators as for the high performance OCXO's. These oscillators, therefore, exhibit a particularly good noise performance and also a superior ageing characteristic.



FS 5901-11

FS 5901-01 & FS 5903

FS 5905 & FS 5906

SPECIFICATION APPLICABLE TO ALL TYPES

Supply Voltage	5V ±5%
Output	Square wave TTL compatible High level 2.4V min. Low level 0.4V max.
Voltage Coefficient	Less than 1 ppm for a supply change of 5%
Load Coefficient	Less than 0.2 ppm open circuit to full load

SPECIFICATION APPLICABLE TO SPECIFIC TYPES

Long Term Stability

FS 5901-01, -03, -04 FS 5903-01, -03 FS 5905-01 FS 5906-01	Typically 2 ppm in first three months, then better than 2 ppm (typically 1 ppm) per year

Frequency Pulling Characteristic

FS 5903-01, -03 FS 5906-01	3 ppm per volt nominal in the range 1V to 6V

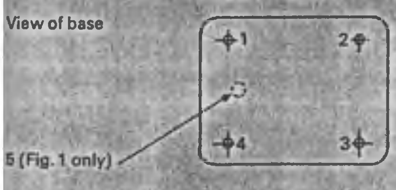
Version FS5903 has frequency control via a voltage applied to pin 4

Version FS5906 has frequency control via a voltage applied to pin 3

Type Number	Frequency Range	Frequency Stability Referred to 25°C		Frequency Accuracy at 25°C	Input Power mW	Max Load	Size
FS 5901-01	5MHz–20MHz	±7.5 ppm OR ±25 ppm	0 to 60°C / −40 to 80°C	External adjustment via trimmer	200	Fan out 2	Fig. 1
FS 5901-04	20MHz–40MHz				200	Fan out 2	Fig. 1
FS 5901-03	300kHz–5MHz				300	Fan out 8	Fig. 1
FS 5901-11	4MHz–20MHz	±50 ppm	0 to 60°C	± 20 ppm	200	Fan out 2	Fig. 2
FS 5901-14	20MHz–40MHz				200	Fan out 2	Fig. 2
FS 5901-13	300kHz–4MHz				300	Fan out 8	Fig. 2
FS 5903-01	5MHz–20MHz	±7.5 ppm OR ±25 ppm	0 to 60°C / −40 to 80°C	External adjustment via trimmer	200	Fan out 2	Fig. 1
FS 5903-03	300kHz–5MHz				300	Fan out 8	Fig. 1
FS 5905-01	5MHz–20MHz	±7.5 ppm OR ±25 ppm	0 to 60°C / −40 to 80°C	±20 ppm	200	Fan out 2	Fig. 3
FS 5905-11	4MHz–20MHz	±50 ppm	0 to 60°C	± 20 ppm	200	Fan out 2	Fig. 3
FS 5906-01	5MHz–20MHz	±7.5 ppm OR ±25 ppm	0 to 60°C / −40 to 80°C	Nominal with 3.5 to 4.5V on pin 3	200	Fan out 2	Fig. 3

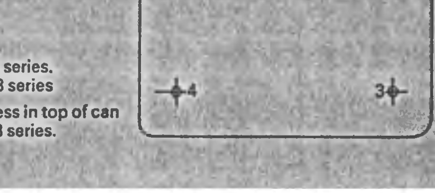
ELECTRICAL CONNECTIONS

FIGS. 1 and 2



1 Output
2 0 volts
3 +5 Volts
4 Not connected for FS 5901 series. Control voltage for FS 5903 series
5 Frequency adjustment access in top of can for FS 5901–0 and FS 5903 series.

FIG. 3



1 Output
2 0 volts
3 Not connected for FS 5905 series Control voltage for FS 5906 series
4 +5 volts

VCXO Range

S 5909
OLTAGE CONTROLLED
RYSTAL OSCILLATOR

he frequency of this source can be aried by the application of voltage to ne pin of the oscillator.

eatures

- Mechanically robust
- Small size and weight
- Hermetically sealed metal enclosure
- Wide frequency pulling range
- Good linearity and stability

FS 5909

OUTLINE SPECIFICATION	FS 5909 SERIES
Frequency Range	5 MHz to 20 MHz
Power Supply Options	5, 8, 9, 12 and 15V ±5%
Power Consumption	250 mW
Max. Operating Temperature Range	−30°C to 80°C
Storage Temperature Range	−55°C to 105°C
Frequency Stability all causes*	±100 ppm −30°C to 105°C ± 50 ppm −25°C to 80°C
Frequency Tolerance on nominal at 25°C (V.C. = 3.5v)	±20 ppm
Frequency Adjustment with Varicap (1V to 6V)	±25 ppm min. < 10MHz ±40 ppm min. > 10MHz
Output	To drive up to ten low power Schottky logic loads

*including temperature, load and voltage variations plus an allowance for ageing over ten years.

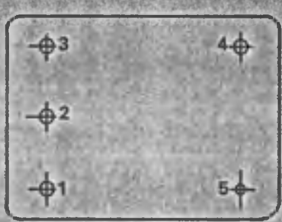
Ordering Information

When ordering please give:

1. Oscillator frequency
2. Supply voltage

LECTRICAL CONNECTIONS

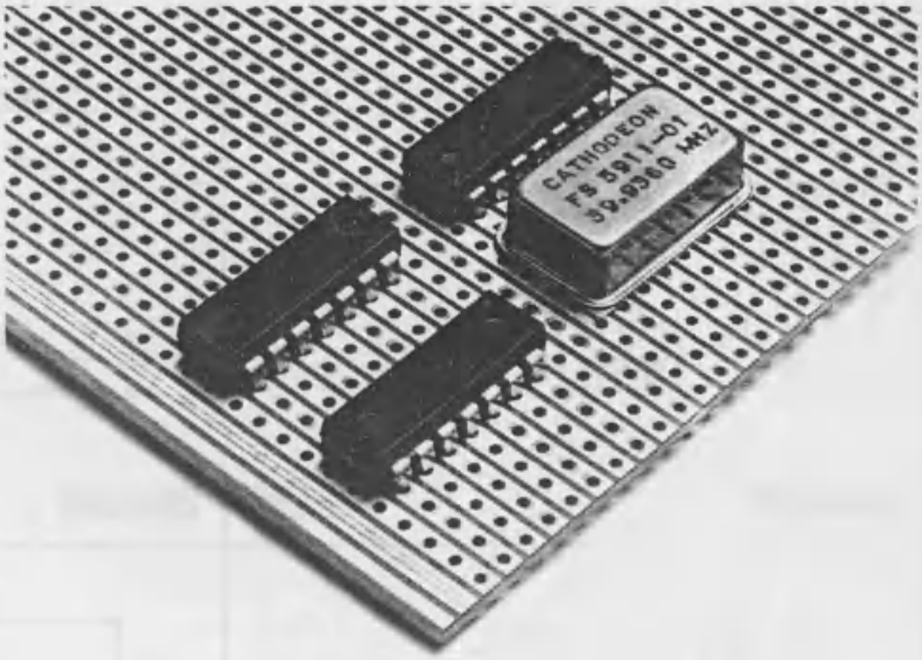
View of base



1 Frequency adjustment
2 Test Point
3 + DC volts
4 RF Output
5 DC and RF common

Clock Oscillators Dual-in-Line (DIL)

RYSTAL CLOCK OSCILLATORS
0 kHz to 70 MHz



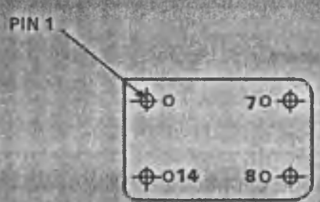
atures
- Low Power Consumption
- Wide Operating Temperature Range
- Small Size, low profile
- Hermetically Sealed Metal Enclosure
- Automatic Assembly
- Guaranteed Performance
- Modular Reliability
- Low Cost

pplications
- Time Bases
- Ruggedized Equipment
- Counters
- System Clocks

ompatibility
- This device has pin spacings compatible with 14 pin DIL packages
- TTL Output

OUTLINE SPECIFICATION	FS 5911 SERIES	FS 5912 SERIES
Frequency Range	4 MHz to 60 MHz	250 kHz to 3.999 MHz 60 MHz to 70 MHz
Power Supply	5V ±0.5V	5V ±0.5V
Supply Current (maximum)	25 mA	40 mA
Max. Operating Temperature Range	0°C to 70°C	0°C to 70°C
Storage Temperature Range	−55°C to 125°C	−55°C to 125°C
Frequency Stability (for all causes)	Version 01 ± 50 ppm 02 ±100 ppm 03 ±500 ppm	Version 01 ± 50 ppm 02 ±100 ppm 03 ±500 ppm
Output — Waveform	TTL	TTL
— Level	'0' = 0.4V Max. '1' = 2.4V Min.	'0' = 0.4V Max. '1' = 2.4V Min.
— Symmetry	50 ±10%	50 ±10%
— Load	1 to 10 TTL Gates	1 to 10 TTL Gates

LECTRICAL CONNECTIONS



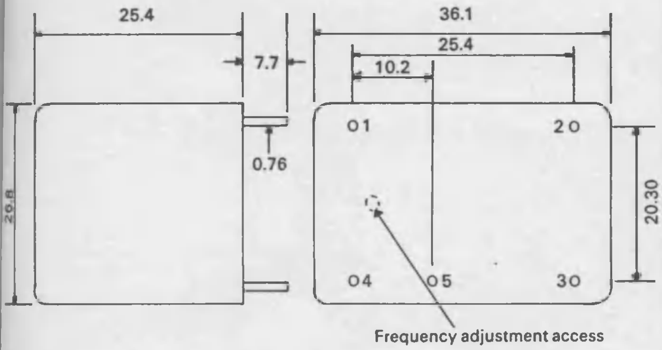
View of base

PIN CONNECTIONS

1	No connection
7	GND
8	Output
14	+5V DC

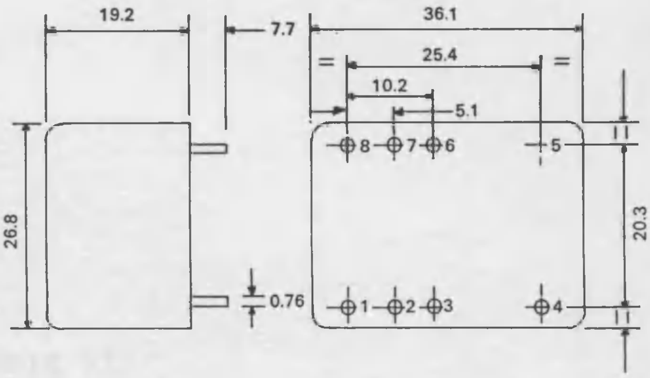
Outline Drawings

Dimensions in mm are maximum except pin centres and diameters which are nominal.



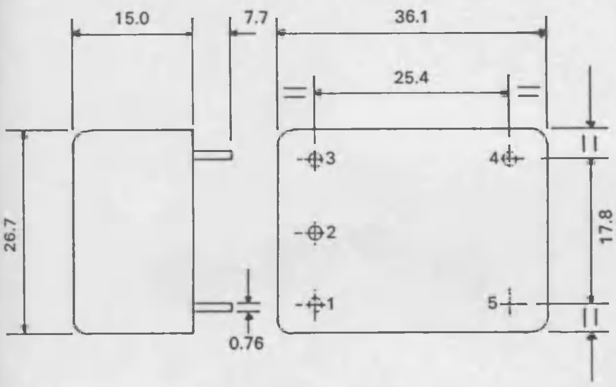
FS 5951 & FS 5953

Package A



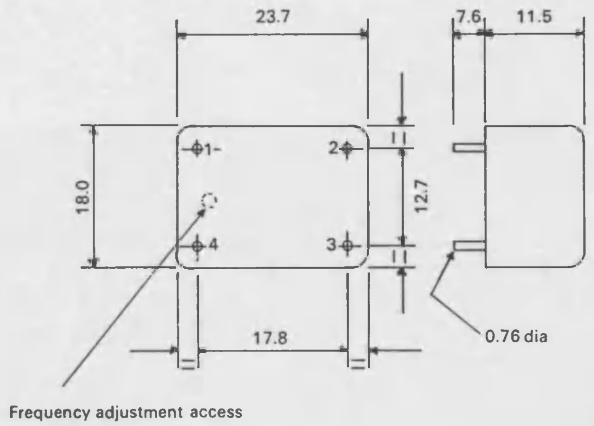
FS 5805

Package B



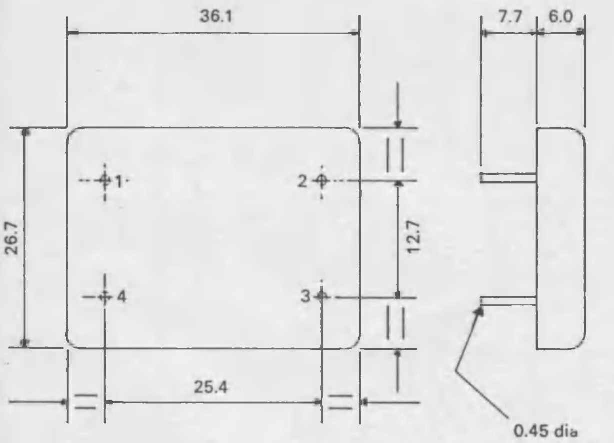
FS 5909 & FS 5815

Package C



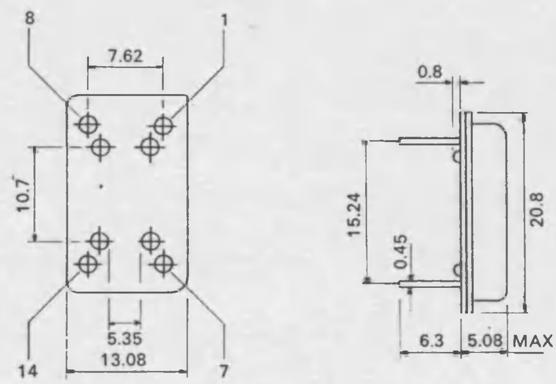
FS 5901 & FS 5903

Package D



FS 5905 & FS 5906

Package E



FS 5911 & FS 5912

Package F

Cathodeon Crystals Ltd Linton Cambridge CB1 6JU England Telephone Cambridge (0223) 891501

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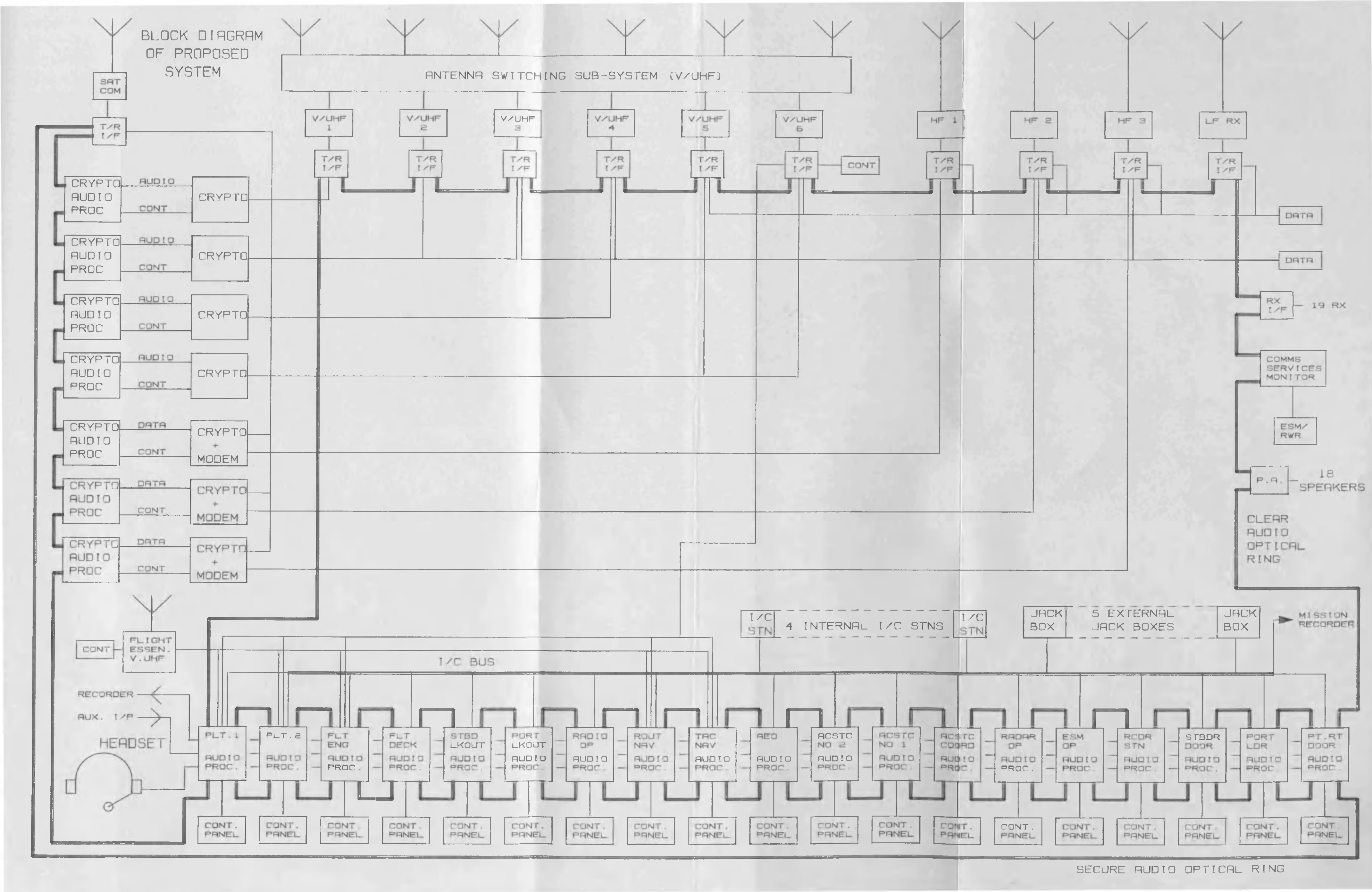
Printed in England

APPENDIX VII

System Block Diagram


BLOCK DIAGRAM OF PROPOSED SYSTEM
SAT COM
T/R I/F
ANTENNA SWITCHING SUB-SYSTEM (V/UHF)
V/UHF 1
V/UHF 2
V/UHF 3
V/UHF 4
V/UHF 5
V/UHF 6
HF 1
HF 2
HF 3
LF RX
CONT
CRYPTO AUDIO PROC
AUDIO
CONT
DATA
CRYPTO
CRYPTO + MODEM
DATA
RX I/F
19 RX
COMMS SERVICES MONITOR
ESM/ RWR
P.A.
18 SPEAKERS
CLEAR AUDIO OPTICAL RING
FLIGHT ESSEN. V.UHF
I/C STN
4 INTERNAL I/C STNS
JACK BOX
5 EXTERNAL JACK BOXES
MISSION RECORDER
I/C BUS
RECORDER
AUX. I/P
HEADSET
PLT.1 AUDIO PROC.
PLT.2 AUDIO PROC.
FLT ENG AUDIO PROC.
FLT DECK AUDIO PROC.
STBD LKOUT AUDIO PROC.
PORT LKOUT AUDIO PROC.
RADIO OP AUDIO PROC.
ROUT NAV AUDIO PROC.
TAC NAV AUDIO PROC.
AEO AUDIO PROC.
ACSTC NO 2 AUDIO PROC.
ACSTC NO 1 AUDIO PROC.
ACSTC COORD AUDIO PROC.
RADAR OP AUDIO PROC.
ESM OP AUDIO PROC.
RCDR STN AUDIO PROC.
STBDR DOOR AUDIO PROC.
PORT LDR AUDIO PROC.
PT.RT DOOR AUDIO PROC.
CONT. PANEL
SECURE AUDIO OPTICAL RING